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(54) **SYSTEM AND METHOD FOR ROUTING A CALL TO A CALLED PARTY'S LANDLINE OR WIRELESS COMMUNICATION UNIT**

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(58) **Field of Search** 455/412, 413, 455/414, 417, 418, 422, 432, 433, 435, 436, 445, 461, 560, 561, 563, 564, 569; 379/201, 207, 210, 211, 212, 88.15, 88.22, 88.23, 88.24, 88.25, 88.26

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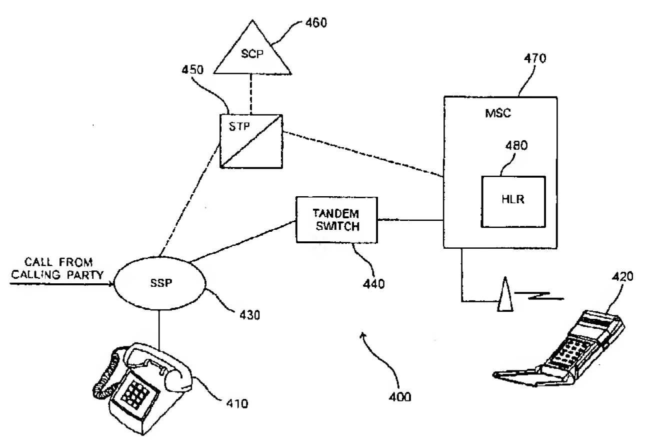
(74) *Attorney, Agent, or Firm*—Brinks Hofer Gilson & Lione

(57) **ABSTRACT**

A method and system are provided for routing a call to a called party's landline or wireless communication unit based on the availability of the wireless communication unit. A call placed to a called party's landline communication unit is routed to the called party's wireless communication unit if the wireless communication unit is determined available, otherwise, the call is routed to the called party's landline communication unit. Unlike past attempts to facilitate communication with a called party having wireless and landline communication units, these preferred embodiments operate automatically (without called party intervention) before a call is terminated at the landline communication unit and are responsive to the availability of the called party's wireless communication unit. Further, the re preferred embodiments do not rely on complex, pre-determined hunting sequences or expensive adjunct customer premises equipment.

20 Claims, 6 Drawing Sheets



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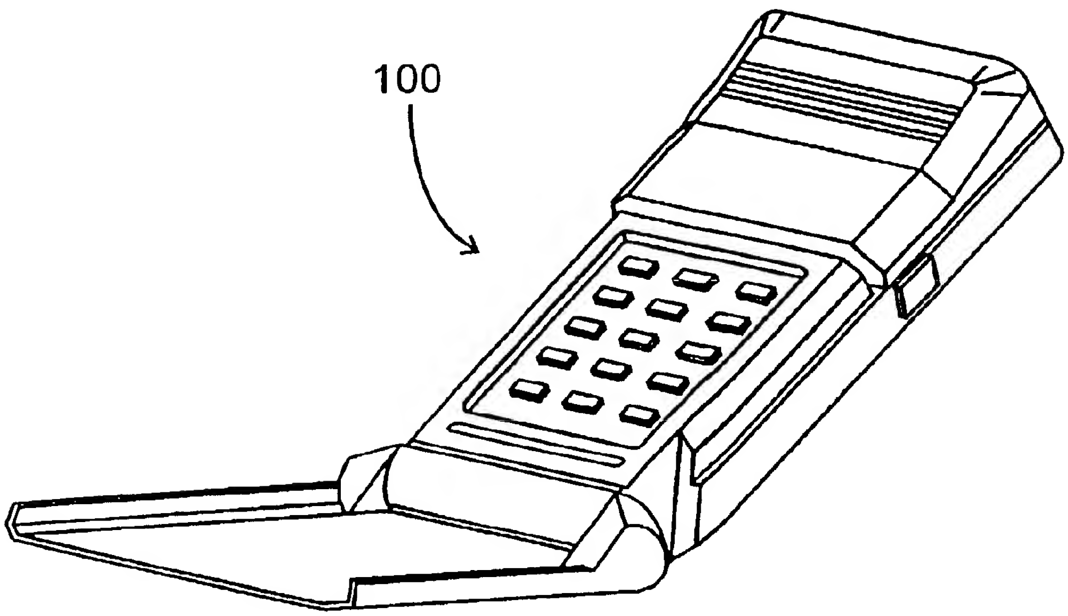


Fig. 1

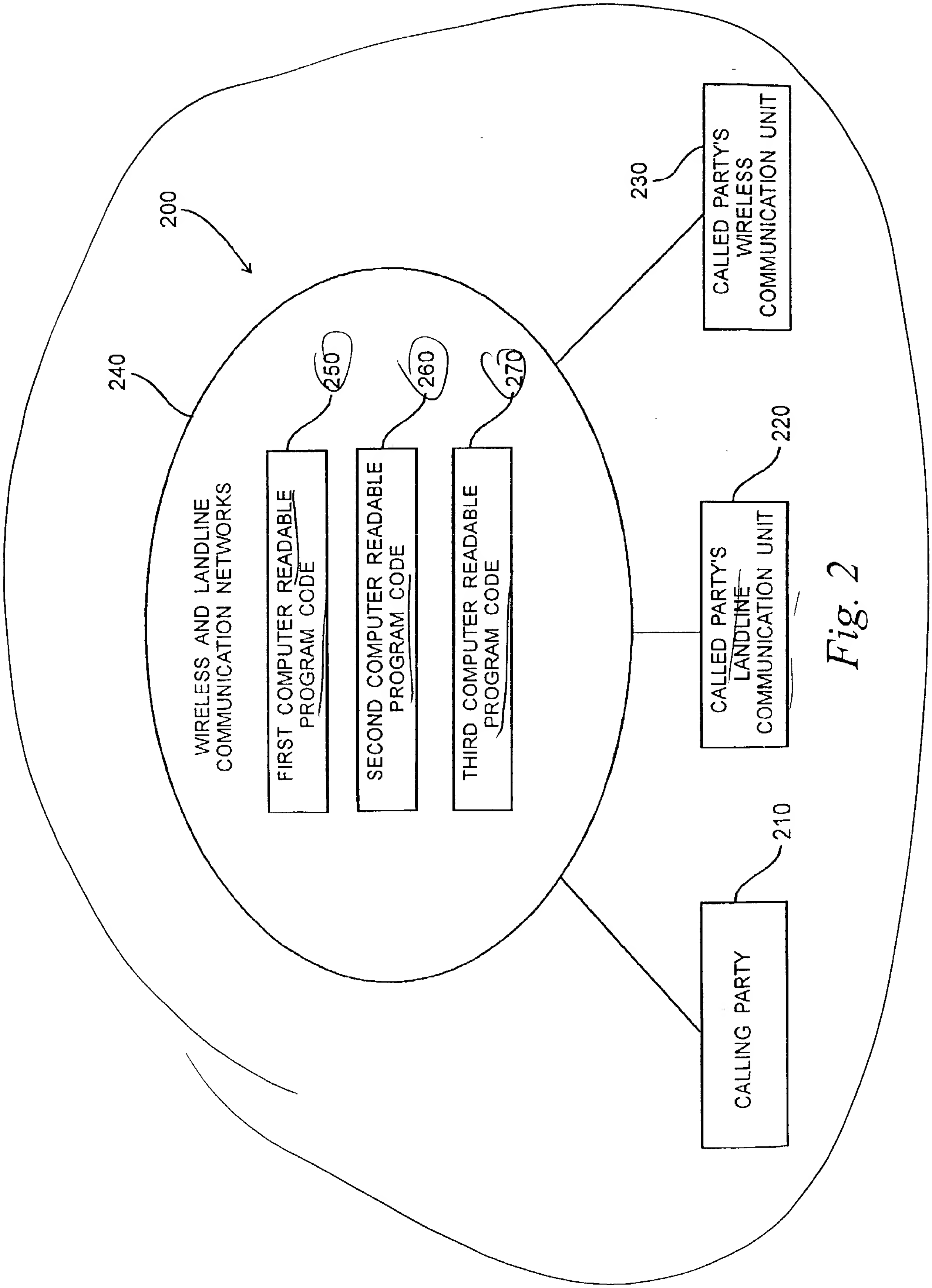

200
240
WIRELESS AND LANDLINE COMMUNICATION NETWORKS
250
FIRST COMPUTER READABLE PROGRAM CODE
260
SECOND COMPUTER READABLE PROGRAM CODE
270
THIRD COMPUTER READABLE PROGRAM CODE
230
CALLED PARTY'S WIRELESS COMMUNICATION UNIT
220
CALLED PARTY'S LANDLINE COMMUNICATION UNIT
210
CALLING PARTY

Fig. 2

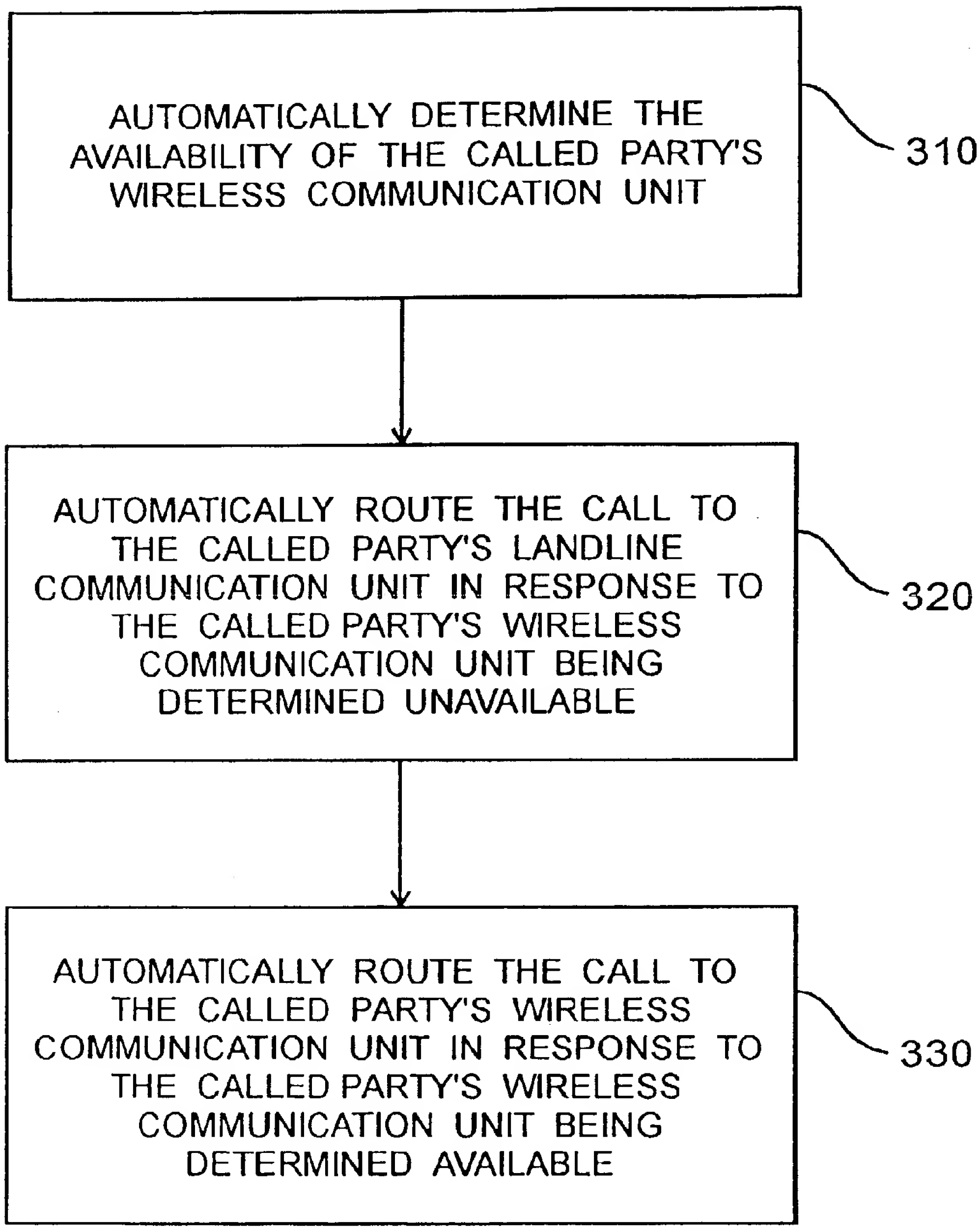
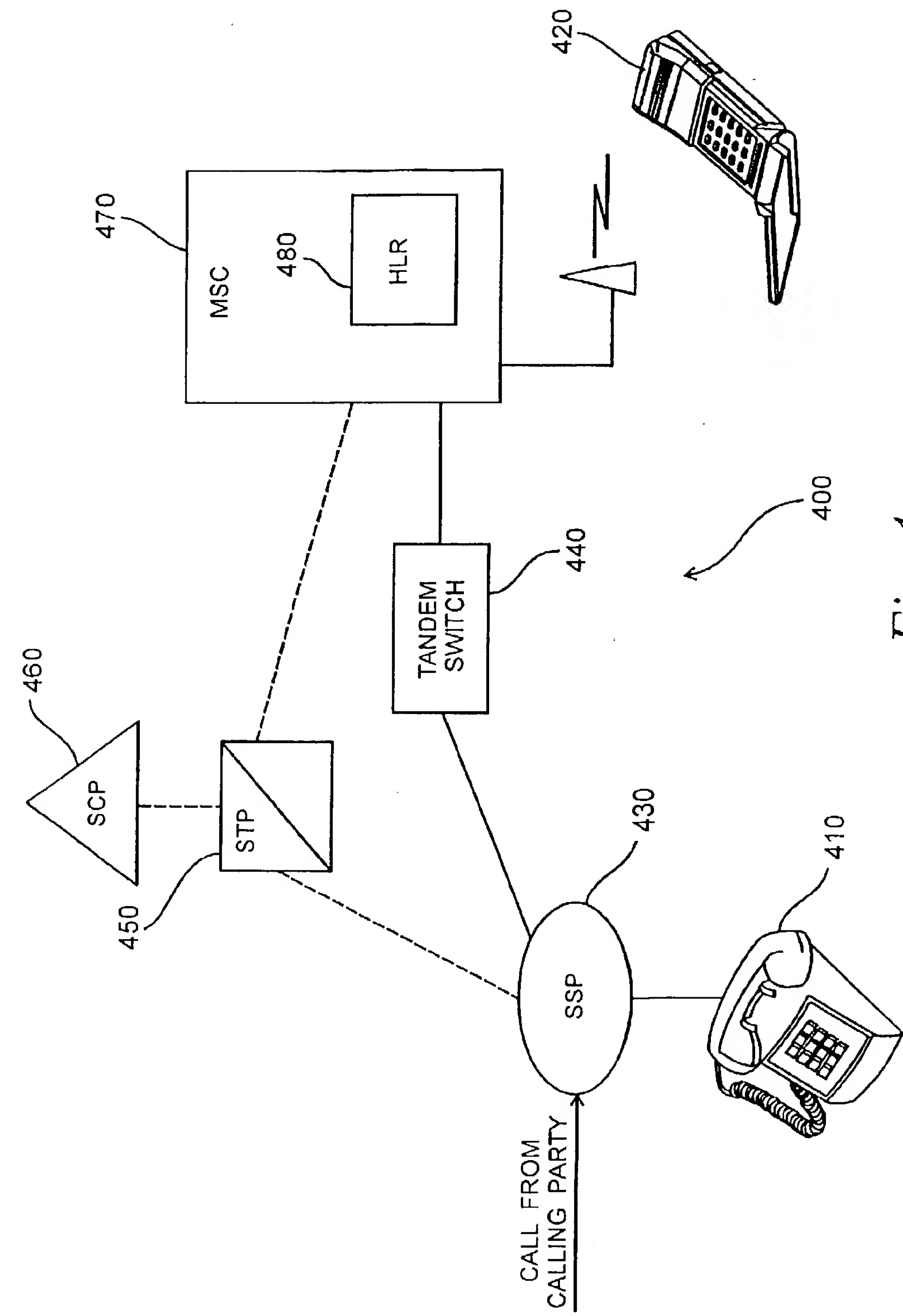

AUTOMATICALLY DETERMINE THE AVAILABILITY OF THE CALLED PARTY'S WIRELESS COMMUNICATION UNIT
310
AUTOMATICALLY ROUTE THE CALL TO THE CALLED PARTY'S LANDLINE COMMUNICATION UNIT IN RESPONSE TO THE CALLED PARTY'S WIRELESS COMMUNICATION UNIT BEING DETERMINED UNAVAILABLE
320
AUTOMATICALLY ROUTE THE CALL TO THE CALLED PARTY'S WIRELESS COMMUNICATION UNIT IN RESPONSE TO THE CALLED PARTY'S WIRELESS COMMUNICATION UNIT BEING DETERMINED AVAILABLE
330

Fig. 3



470
MSC
480
HLR
420
TANDEM SWITCH
440
400
460
SCP
STP
450
430
SSP
410
CALL FROM CALLING PARTY

Fig. 4

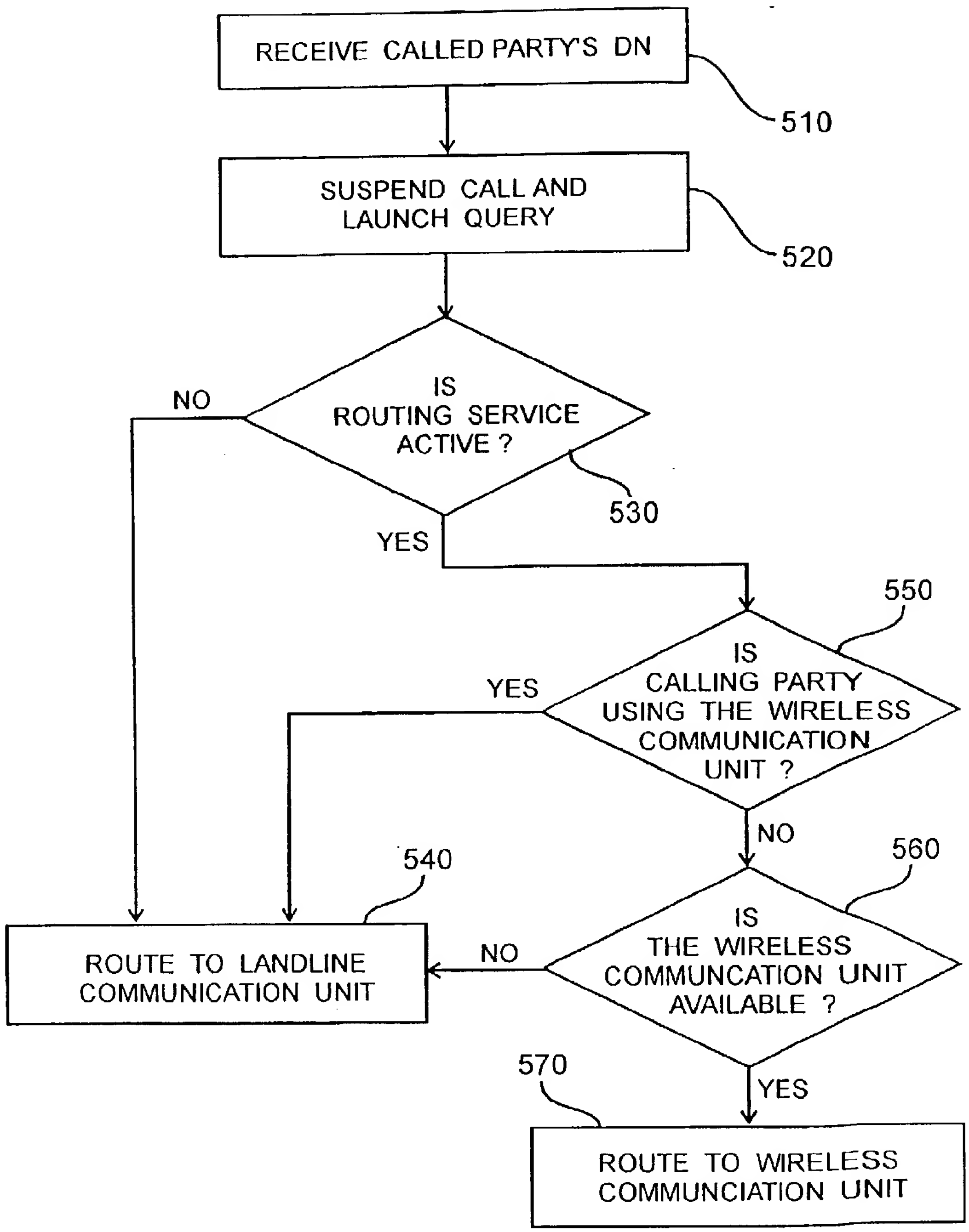

RECEIVE CALLED PARTY'S DN
510
SUSPEND CALL AND LAUNCH QUERY
520
IS ROUTING SERVICE ACTIVE ?
530
NO
YES
550
IS CALLING PARTY USING THE WIRELESS COMMUNICATION UNIT ?
YES
NO
540
560
ROUTE TO LANDLINE COMMUNICATION UNIT
NO
IS THE WIRELESS COMMUNCATION UNIT AVAILABLE ?
570
YES
ROUTE TO WIRELESS COMMUNCIATION UNIT

Fig. 5

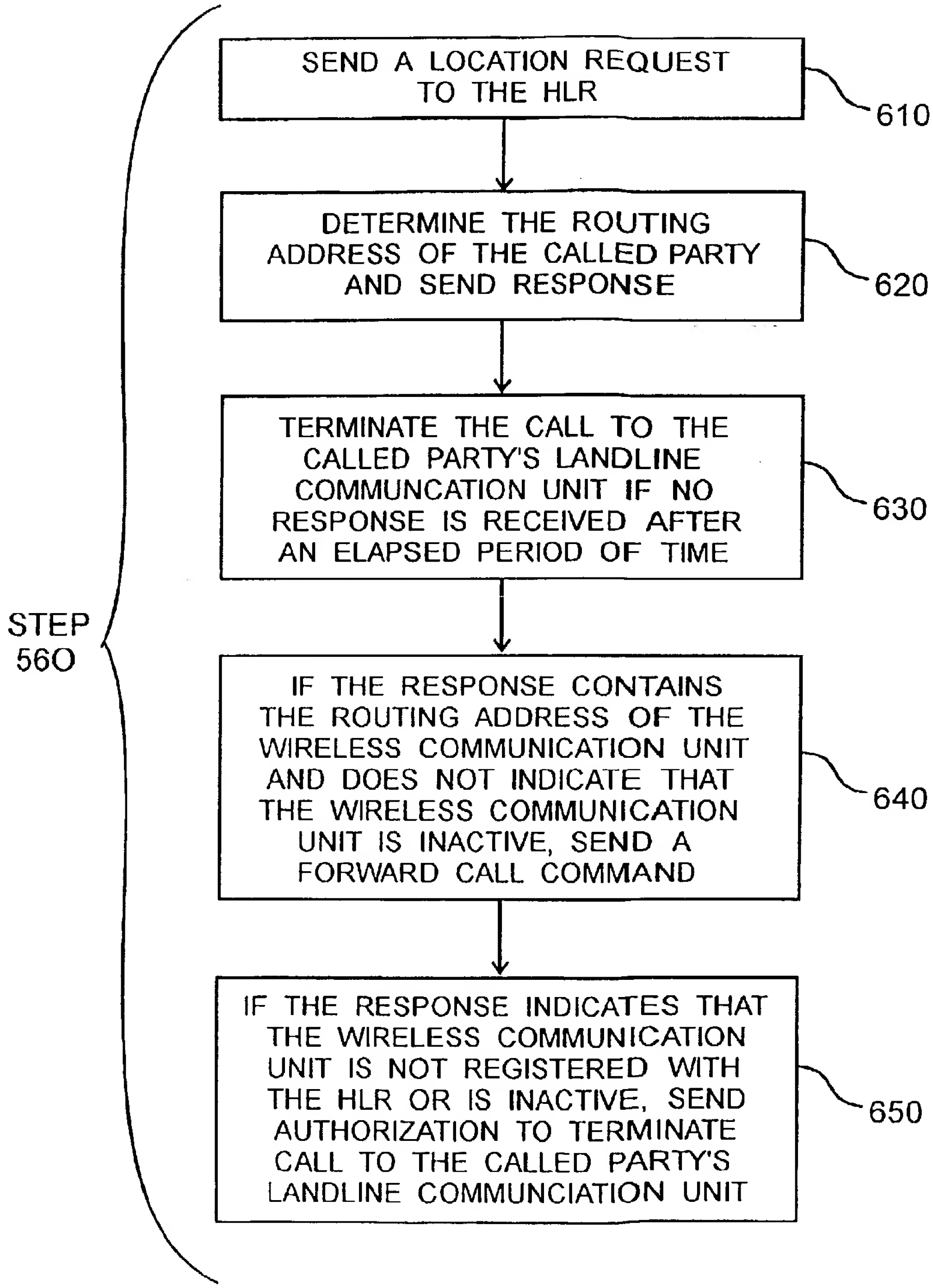

SEND A LOCATION REQUEST TO THE HLR
610
DETERMINE THE ROUTING ADDRESS OF THE CALLED PARTY AND SEND RESPONSE
620
TERMINATE THE CALL TO THE CALLED PARTY'S LANDLINE COMMUNCATION UNIT IF NO RESPONSE IS RECEIVED AFTER AN ELAPSED PERIOD OF TIME
630
IF THE RESPONSE CONTAINS THE ROUTING ADDRESS OF THE WIRELESS COMMUNICATION UNIT AND DOES NOT INDICATE THAT THE WIRELESS COMMUNICATION UNIT IS INACTIVE, SEND A FORWARD CALL COMMAND
640
IF THE RESPONSE INDICATES THAT THE WIRELESS COMMUNICATION UNIT IS NOT REGISTERED WITH THE HLR OR IS INACTIVE, SEND AUTHORIZATION TO TERMINATE CALL TO THE CALLED PARTY'S LANDLINE COMMUNCIATION UNIT
650
STEP 560

Fig. 6

Call forwarding / Routing to called party

SYSTEM AND METHOD FOR ROUTING A CALL TO A CALLED PARTY'S LANDLINE OR WIRELESS COMMUNICATION UNIT

TECHNICAL FIELD

The present invention relates generally to call processing in telecommunication networks and specifically to a system and method for routing a call to a called party's landline or wireless communication unit.

BACKGROUND

Many people have a wireless communication unit, such as a cellular phone, in addition to a landline communication unit, such as a home telephone. While having a home phone and a cellular phone allows a person to place calls both at and away from home, two phones with unique phone numbers can make reaching the person difficult. For example, a person carrying an active cellular phone will miss a call placed to his home phone.

Several methods have been suggested to facilitate communication with users having wireless and landline communication units. In one method, a user forwards calls from his home phone to his cellular phone by manually entering the cellular phone number into a call forwarding service. After entry of the cellular phone number, all calls to the user's home phone are forwarded from the home phone to the user's cellular phone, even if the cellular phone is inactive. In another method, if a call placed to a user's home phone is not answered after several rings, the call is transferred to the user's cellular phone. Because the call is transferred typically after four or five rings, some calling parties, believing that the call will not be answered, hang up before the call is transferred. Other methods rely on complex, pre-determined hunting sequences or expensive adjunct customer premises equipment.

There is a need, therefore, for an improved system and method for routing a call to a called party's landline or wireless communication unit that will overcome the disadvantages described above.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is an illustration of a wireless communication unit of a preferred embodiment.

FIG. 2 is a block diagram of a telecommunication system of a preferred embodiment.

FIG. 3 is a flow chart of a method of a preferred embodiment for automatically routing a call to a called party's landline or wireless communication unit.

FIG. 4 is a block diagram of one preferred embodiment of the telecommunication system of FIG. 2.

FIG. 5 is a flow chart of a method of a preferred embodiment for automatically routing a call to a called party's landline or wireless communication unit using the telecommunication system of FIG. 4.

FIG. 6 is a flow chart of a method of a preferred embodiment for determining whether a called party's wireless communication unit is available.

DETAILED DESCRIPTION OF THE PRESENTLY PREFERRED EMBODIMENTS

By way of introduction, the preferred embodiments described below include a method and system for routing a call to a called party's landline or wireless communication unit based on the availability of the wireless communication unit. In one embodiment, a call placed to a called party's landline communication unit is routed to the called party's wireless communication unit, such as a cellular phone 100 (FIG. 1), if the wireless communication unit is available. Otherwise, the call is routed to the called party's landline communication unit. Unlike past attempts to facilitate communication with a called party having wireless and landline communication units, these preferred embodiments operate automatically (without called-party intervention) before a call is terminated at the landline communication unit and are responsive to the availability of the called party's wireless communication unit. Further, there preferred embodiments do not rely on complex, pre-determined hunting sequences or expensive adjunct customer premises equipment.

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Turning again to the drawings, FIG. 2 is a block diagram of a telecommunication system 200 of a preferred embodiment. As shown in FIG. 2, a calling party 210 is coupled with a called party's landline communication unit 220 and wireless communication unit 230 through wireless and landline communication networks 240. As used herein, the term "coupled with" means directly coupled with or indirectly coupled with through one or more components. The wireless and landline communication networks 240 comprise computer usable medium having first, second, and third computer readable program codes 250, 260, 270 embodied therein. It is important to note that while the program codes 250, 260, 270 have been shown as three separate components, their functionality can be combined and/or distributed. It is also important to note that "media" is intended to broadly cover any suitable media, analog or digital, now in use or developed in the future.

1-3 Readable program code

The telecommunication system 200 of FIG. 2 can be used in a method for automatically routing a call to a called party's landline or wireless communication unit, as shown in the flow chart of FIG. 3. When the calling party 210 places a call to the called party, the first computer readable program code 250 automatically determines the availability of the called party's wireless communication unit 230 (step 310). In response to the called party's wireless communication unit 230 being determined unavailable, the second computer readable program code 260 automatically routes the call to the called party's landline communication unit 220 (step 320). In response to the called party's wireless communication unit 230 being determined available, the third computer readable program code 270 automatically routes the call to the called party's wireless communication unit 230 (step 330).

FIG. 4 is a block diagram of one preferred embodiment 400 of the telecommunication system 200 of FIG. 2. This preferred system 400 comprises a landline communication unit 410 coupled with a wireless communication unit 420 through a signal switching point (SSP) 430, a tandem switch 440, a signal transfer point (STP) 450, a service control point (SCP) 460, a mobile switching center (MSC) 470, and a home location register (HLR) 480. This system 400 embodies an intelligent bridge between components used in wireless and landline networks. The MSC 470 and the HLR 480 are components used in a wireless network. The MSC 470 handles switching and routing to the wireless communication unit 420, and the HLR 480 is a database that stores information about the location of the wireless communication unit 420.

The other components of the system 400 communicate voice and data traffic and network signaling protocols that control switching of the voice and data traffic. The SSP 430 is a central office equipped with Advanced Intelligent Network (AIN) software, which enables the SSP 430 to suspend

call processing and launch a query to the SCP **460** via the STP **450**. The SCP **460** handles queries sent from the SSP **430** by communicating with HLR **480**, although any database that contains the information described below can be used. Preferably, communication between the SCP **460** and the HLR **480** is carried out through a Signaling System 7 (SS7) network using IS-41 Transaction Capabilities Applications Protocol (TCAP) Rev. B protocol. The SSP **430** also communicates voice and data traffic to the MSC **470** via the tandem switch **440**.

Terminating

voice/data

In this embodiment, the SSP **430** is associated with the called party's landline communication unit **410** and receives a call from a calling party. Because the SSP **430** is associated with the called party, it is referred to as a terminating SSP. It should be understood that an SSP associated with the calling party (an originating SSP) can be modified to perform the functions described below. It is important to note that the SSP **430** can transfer voice and data traffic directly without the use of the tandem switch **440** and can directly transfer network signaling protocols to the SCP **460** without the use of the STP **450**. Also, a central office not equipped with an SSP can be provided with software to send messages to the SCP **460** in an AIN-query format. Preferably, the SCP **460** utilizes a service order interface to create an automated message account (AMA) billing record to charge for calls that are routed using the method described below.

Terminating

FIG. 5 is a flow chart of a method of a preferred embodiment for automatically routing a call to a called party's landline or wireless communication unit **410**, **420** using the preferred system **400** of FIG. **4**. First, the SSP **430** receives the called party's destination number (DN) from the calling party (step **510**). In this preferred embodiment, the called party's DN is the DN associated with the called party's landline communication unit **410**. A terminating attempt trigger (TAT) at the SSP **430** recognizes the called party's DN, and the SSP **430** suspends the call and launches a terminating attempt query to the SCP **460** (step **520**). Preferably, this query follows the form shown in Appendix A.

messages termination and call routing

Next, the SCP **460** determines whether the wireless/landline routine service is active (step **530**). If the wireless/landline routing service is deactive, the SCP **460** responds to the query from the SSP **430** with a message authorizing termination to the called party's landline communication unit **410** (preferably in the form shown in Appendix B), and the SSP **430** routes the call to the landline communication unit **410** (step **540**). If the wireless/landline call routing service is active, the SCP **460** determines whether the calling party is using the wireless communication unit **420** (step **550**). This determination can be made, for example, by analyzing the DN of the calling party identification parameter in the AIN query. If the calling party is using the wireless communication unit **420**, it is preferred that the call be routed to the called party's landline communication unit **410** (step **540**) to avoid automatically routing the call back to the wireless communication unit **420** itself (i.e., to prevent circular routing).

determination

If the calling party is not using the wireless communication unit **420** (or if the SCP **460** cannot make the determination), the SCP **460** determines whether the called party's wireless communication unit **420** is available (step **560**). The preferred execution of this step is shown in the flow chart of FIG. **6**. First, SCP **460** sends a location request to the HLR **480** of the MSC **470** (step **610**). It is preferred that the location request follow the form shown in Appendix C. Upon receiving the request the HLR **480** determines the routing address of the wireless communication unit **420** and sends a response containing the routing address, if available, to the SCP **460** (step **620**). If no response is received by the SCP **460** after an elapsed period of time, the SCP **460** preferably authorizes the SSP **430** to terminate the call to the called party's landline communication unit **410** (step **630**). If the response contains the routing address of the wireless communication unit **420** and does not indicate that the wireless communication unit **420** is inactive (erg., if the "access denied" field is populated by any value except "2" (inactive)), the SCP **460** sends a forward call command to the SSP **430** (step **640**), and the call is routed through the MSC **470** to the wireless communication unit **420** (step **570**). A preferred response format from the HLR **480** is shown in Appendix D, and a preferred forward call command format is shown in Appendix E.

If the response indicates that the wireless communication unit **420** is not registered with the HLR **480** or is inactive (see the preferred response in Appendix F), the SCP **460** sends the SSP **430** a message authorizing termination to the called party's landline communication unit **410**, preferably in the form shown in Appendix G (step **650**). The SSP **430** then routes the call to the landline communication unit **410** (step **540**).

forwarded to another number/ voice mail

It is possible that a call routed to an active wireless communication unit **420** will go unanswered. Such a call can be handled by the logic in the MSC **470** (e.g., termination of the call, transfer to voice mail, call forward to another number, etc.). To prevent circular routing of calls, it is preferred that the calling party DN be changed to the DN of the wireless communication unit **420** for calls routed to the wireless communication unit **420**. In this way, if, for example, the MSC **470** is programmed to forward unanswered calls to the landline communication unit **410**, the automatic landline/routing feature described above will not re-route the call to the wireless communication unit **420** (see step **550** above).

programmed to forward

Busy Line

It is also possible that a call routed to an active wireless communication unit **420** will terminate on a busy line. As described above, the MSC **470** can transfer such a call to the voice-mail system associated with the wireless communication unit **420**. A called party, however, may not be aware of the need to check that voice-mail system. To avoid this problem, a unified voice-mail system can be used to handle calls from both the wireless communication unit **420** and the landline communication unit **410**. In this way, a user can check the voice-mail system associated with the landline communication unit **410** and receive messages left by callers who were routed to the wireless communication unit **420**.

In the above-described preferred embodiment, the availability of the wireless communication unit **420** was based on whether the wireless communication unit **420** was active. In an alternative embodiment, availability is additionally based on whether the wireless communication unit **420** is busy. In this embodiment, the SCP **460** authorizes the SSP **430** to route the call to the wireless communication unit **420** only if the unit **420** is both active and not busy.

While the number of the called party's landline communication unit **410** was used in the above-described preferred embodiments to trigger a query, a number not associated with either the called party's landline or wireless communication unit can be used to trigger the query. Also, a query can be triggered when a calling party dials the number of the called party's wireless communication unit **420**. As one example of this alternative embodiment, a call to the placed wireless communication unit **420** can be automatically forwarded by the MSC **470** to the SSP **430**, which launches a

query as described above. Of course, if a call is placed to the non-triggering number, the call will be directly connected to that communication unit. For example, in the embodiment illustrated in FIG. **5**, a call to the wireless communication unit **420** will be directly connected to that unit **420**.

In another alternative embodiment, if a call is routed to the communication unit that is not associated with the number called by the calling party, an indication can be generated by that communication unit. For example, if a calling party dials a number for the called party's landline communication unit and if the call is routed to the called party's active wireless communication unit, a distinctive ring, such as a multiple ring, can be generated by the wireless communication unit. In contrast, a call directly terminated to the wireless communication unit can have a single ring. In this way, a called party can monitor the ring pattern of a communication unit to determine whether the incoming call is a direct or routed call. Of course, other kinds of indications~~, such as a visual indication, can be generated.~~

Voice Response

As mentioned in the discussion above with reference to FIG. **5**, it is possible to deactivate the wireless/landline call routing service. One way of engaging or disengaging the service is through an interactive voice response (IVR) unit. A called party can disengage the service if ~~he does not want~~ to be disturbed by or incur the expense of a routed call. One way in which to deactivate the wireless/landline call routing service is to engaging an unconditional call forwarding feature.

As another alternative, the SCP **460** can instruct the SSP **430** to play a message to the calling party while waiting for a response from the HLR **480**. This message can alert the calling party that the call is being processed, providing the advantage of alerting the calling party that the call is being processed despite a delay, which can occur, for example, if the called party is roaming between MSCs.

For simplicity, the term "landline communication unit" is intended to broadly cover any communication unit that receives calls from a calling party through a physical connection from a main switch point, such as a central office. Landline communication units include, but are not limited to, home or office telephones, fax machines, and modems. Also for simplicity, the term "wireless communication unit" is intended to broadly cover any communication unit that receives calls from a calling party through a wireless, over-the-air connection. Wireless communication units preferably include, but are not limited to, cellular phones, mobile phone, paging devices, and modems adapted to receive wireless transmissions, although personal communication service (PCS) devices can also be used. Wireless communication units can use any wireless communication technology including, but not limited to, analog with enhanced registration, time division multiple access (TDMA), code division multiple access (CDMA), and global system multiple (GSM) technology, as well as radio, infrared, and satellite transmissions.

It is intended that the foregoing detailed description be understood as an illustration of selected forms that the invention can take and not as a definition of the invention. It is only the following claims, including all equivalents, that are intended to define the scope of this invention.

Appendix A

AIN Query
- Termination Attempt
- Called Party ID—Called Number
- Lata—LATA ID
- Calling Party ID—Calling Number (if available)
- Original Called Party ID (if available)
- Redirected Party ID (if available)
- Redirection Information (if available)

Appendix B

AIN Response
- Auth_Term
- CallingPartyID=Calling Number (if available)
- CalledPartyID=Landline Number (as dialed)

Appendix C

IS-41 INVOKE
- LOCREQ
- Digits(dialed)
- MSCID
- SystemMyTypeCode
- Billing ID

Appendix D

IS-41 RETURN RESULT
- locreq
- MSCID
- MIN
- MSD
- Digits—Routing Address

Appendix E

AIN Response
- Forward Call
- CallingPartyID=Calling Number (if available)
- CallingPartyID=Wireless Number
- Primary Carrier
- AMAAlernatBillingNumber=Wireless Number
- AMAslpID=value from wireless service provider

Appendix F

IS-41 RETURN RESULT
- locreq
- MSCID
- MIN
- MSD
- AccessDeniedReason=2 (inactive)

Appendix G

AIN Response
- Auth_Term

What is claimed is:

1. A method for routing a call to a called party's landline or wireless communication unit, said method comprising:
   (a) receiving, from a calling party, a destination number assigned to the called party's landline communication units; then
   (b) automatically determining an availability of the called party's wireless communication unit; then
   (c) automatically routing the call to the called party's landline communication unit in response to the called party's wireless communication unit being determined unavailable in (b); and
   (d) automatically routing the call to the called party's wireless communication unit in response to the called party's wireless communication unit being determined available in (b).

2. The method of claim 1 further comprising automatically routing the call to the called party's landline communication unit in response to a calling party using the called party's wireless communication unit.

3. The method of claim 1, wherein (b) comprises

(b1) sending a location request to a home location register; and

(b2) analyzing a response to the location request.

4. The method of claim 1, wherein the called party's wireless communication unit is unavailable if inactive and is available if active.

5. The method of claim 1, wherein the called party's wireless communication unit is unavailable if busy and is available if not busy.

6. The method of claim 1 further comprising providing the called party's wireless communication unit with an indication that the call is automatically being routed to the called party's wireless communication unit and is not a direct call to the wireless communication unit.

7. The method of claim 1 further comprising providing the called party's wireless communication unit with a distinctive ring indicating that the call is not a direct call to the wireless communication unit in response to automatically routing the call to the called party's wireless communication unit.

8. The method of claim 1 further comprising playing a message to a calling party before the call is automatically routed, the message alerting the calling party that the call is being processed.

9. The method of claim 1 further comprising using an interactive voice response system to disable performance of (b)–(d).

10. The method of claim 1 further comprising using a unified voice-mail system operative to receive messages for both the landline communication unit and the wireless communication unit.

11. A method for routing a call to a called party's landline or wireless communication unit, said method comprising:

(a) with a signal switching point (SSP), receiving, from a calling party, a destination number assigned to the called party's landline communication unit;

(b) with the SSP, suspending the call and launching a query to a service control point (SCP);

(c) sending a location request from the SCP to a home location register (HLR) of a mobile switching center (MSC);

(d) sending a response from the MHLR to the SCP;

(e) if the response comprises a routing address of the called party's wireless communication unit and does not indicate that the wireless communication unit is inactive:

(e1) sending a forward call command from the SCP to the SSP;

(e2) forwarding the call from the SSP to the MSC; and

(e3) routing the call through the MSC to the called party's wireless communication unit; and

(f) if the response indicates that the called party's wireless communication unit is not registered with the HLR or is inactive:

(f1) sending a message from the SCP to the SSP authorizing termination to the called party's landline communication unit; and

(f2) routing the call through the SSP to the called party's landline communication unit.

12. The invention of claim 11 further comprising:

with the SSP, receiving, from the calling party, a destination number of the calling party; and

if the destination number of the calling party is the same as the destination number assigned to the called party's wireless communication unit, routing the call to the called party's landline communication unit.

13. The invention of claim 11 further comprising:

if a response from the HLR is not received by the SCP after an elapsed period of time, sending a message from the SCP to the SSP authorizing the SSP to terminate the call to the called party's landline communication unit.

14. The invention of claim 11, wherein a forward call command is sent from the SCP to the SSP in act (e1) only if the called party's wireless communication unit is not busy.

15. The invention of claim 11 further comprising:

if the called party's wireless communication unit is busy, routing the call to a unified voice-mail system operative to receive messages for both the called party's landline communication unit and the called party's wireless communication unit.

16. The invention of claim 11 further comprising:

sending a message from the SCP to the SSP authorizing the SSP to play an announcement to the calling party, the message alerting the calling party that the call is being processed.

17. The invention of claim 11 further comprising disabling performance of acts (c)–(f) using an interactive voice response system.

18. The invention of claim 11 further comprising:

providing the called party's wireless communication unit with an indication that the call is automatically being routed to the called party's wireless communication unit and is not a direct call to the wireless communication unit.

19. The invention of claim 11 further comprising:

if the call routed to the called party's wireless communication unit in (e2) is not answered, changing the destination number of the calling party to the destination number of the called party's wireless communication unit.

20. A method for routing a call to a called party's landline or wireless communication unit using a wireless/landline routing service, said method comprising:

(a) with a signal switching point (SSP), receiving, from a calling party, a destination number assigned to the called party's landline communication unit;

(b) with the SSP, receiving, from the calling party, a destination number of the calling party;

(c) if the destination number of the calling party is the same as the destination number assigned to the called party's wireless communication unit, routing the call to the called party's landline communication unit;

(d) if the destination number of the calling party is not the same as the destination number assigned to the called party's wireless communication unit, with the SSP, suspending the call and launching a query to a service control point (SCP);

(e) with the SCP, determining whether the wireless/landline routing service is active;

(f) if the wireless/landline routing service is not active, sending, from the SCP to the SSP, a message authorizing termination to the called party's landline communication unit and, with the SSP, routing the call to the landline communication unit;

(g) sending a location request from the SCP to a home location register (HLR) of a mobile switching center (MSC);

(h) if a response from the HLR is not received by the SCP after an elapsed period of time, sending a message from the SCP to the SSP authorizing the SSP to terminate the call to the called party's landline communication unit;

(i) sending a response from the HLR to the SCP;

(j) if the response comprises a routing address of the called party's wireless communication unit and does not indicate that the wireless communication unit is inactive:

(j1) sending a forward call command from the SCP to the SSP;

(j2) forwarding the call from the SSP to the MSC; and

(j3) routing the call through the MSC to the called party's wireless communication unit;

(k) if the response indicates that the called party's wireless communication unit is not registered with the HLR or is inactive:

(k1) sending a message from the SCP to the SSP authorizing termination to the called party's landline communication unit; and

(k2) routing the call through the SSP to the called party's landline communication unit;

(i) providing the called party's wireless communication unit with a ringing signal that is different from a ringing signal generated when a call is placed directly to the wireless communication unit;

(m) if the called party's wireless communication unit is busy, routing the call to a unified voice-mail system operative to receive messages for both the called party's landline communication unit and the called party's wireless communication unit; and

(n) if the call routed to the called party's wireless communication unit is not answered, changing the destination number of the calling party to the destination number of the called party's wireless communication unit.

* * * * *

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(12) **United States Patent**
Lamarque, III

(10) **Patent No.:** US 6,674,746 B1
(45) **Date of Patent:** Jan. 6, 2004

(54) **METHOD AND APPARATUS FOR VOICE OVER INTERNET PROTOCOL SWAPPING IN A COMMUNICATIONS SYSTEM**

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(73) Assignee: **Nortel Networks Limited**, St. Laurent (CA)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **09/359,538**

(22) Filed: **Jul. 22, 1999**

(51) **Int. Cl.**^{7} **H04L 12/66**

(52) **U.S. Cl.** **370/352**; 370/466

(58) **Field of Search** 370/260, 352, 370/353, 354, 238, 356, 469, 466; 709/204; 379/211.02, 212.01, 218.01

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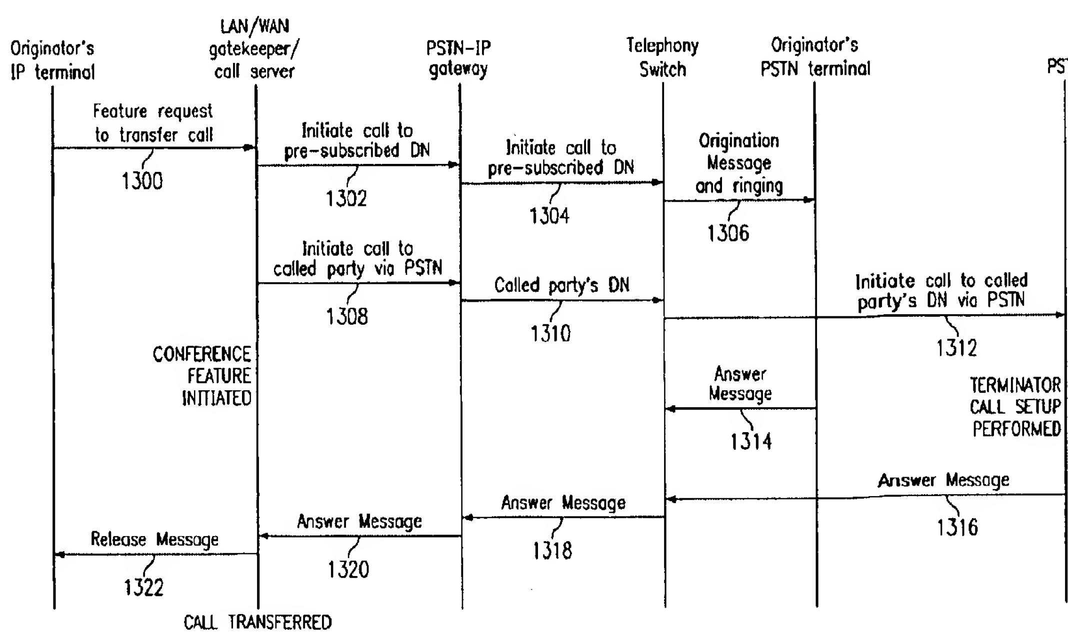
* cited by examiner

Primary Examiner—Chau Nguyen
Assistant Examiner—Nittaya Juntima

(57) **ABSTRACT**

A method and apparatus in a communications system for routing a call. A request is received from a user, at a first terminal, during a call to switch the call from a packet based network to a circuit switched network. Responsive to receiving the request, the call is switched to a second terminal associated with the user, wherein the second terminal uses the circuit switched network and wherein the call is switched to the second terminal without terminating the call.

25 Claims, 8 Drawing Sheets



02/02/2004, EAST Version: 1.4.1

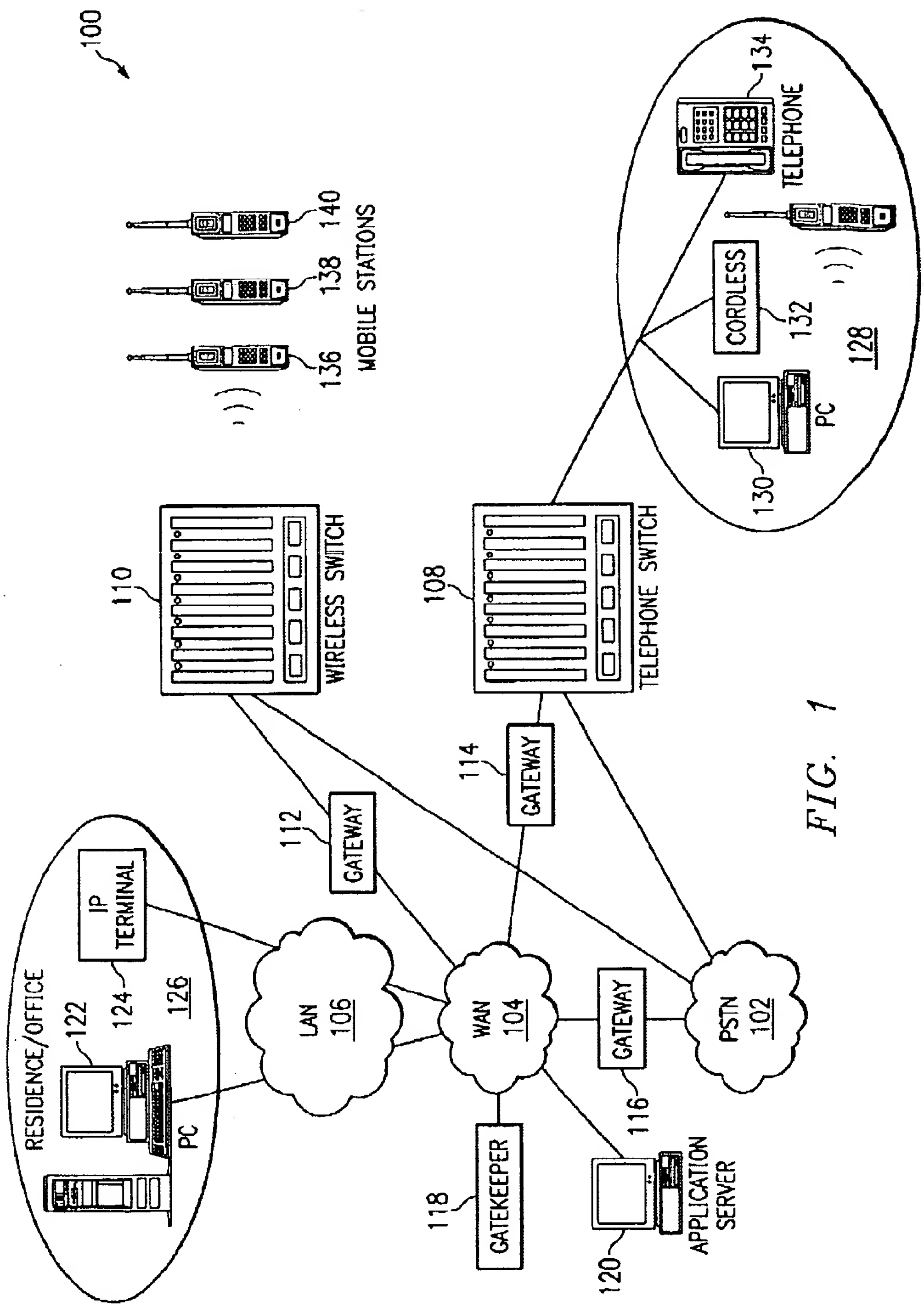

100
RESIDENCE/OFFICE
122
PC
IP TERMINAL
124
126
LAN 106
WAN 104
PSTN 102
GATEWAY
116
GATEKEEPER
118
APPLICATION SERVER
120
GATEWAY
112
GATEWAY
114
WIRELESS SWITCH
110
TELEPHONE SWITCH
108
136
138
140
MOBILE STATIONS
PC
130
CORDLESS
132
128
TELEPHONE
134

FIG. 1

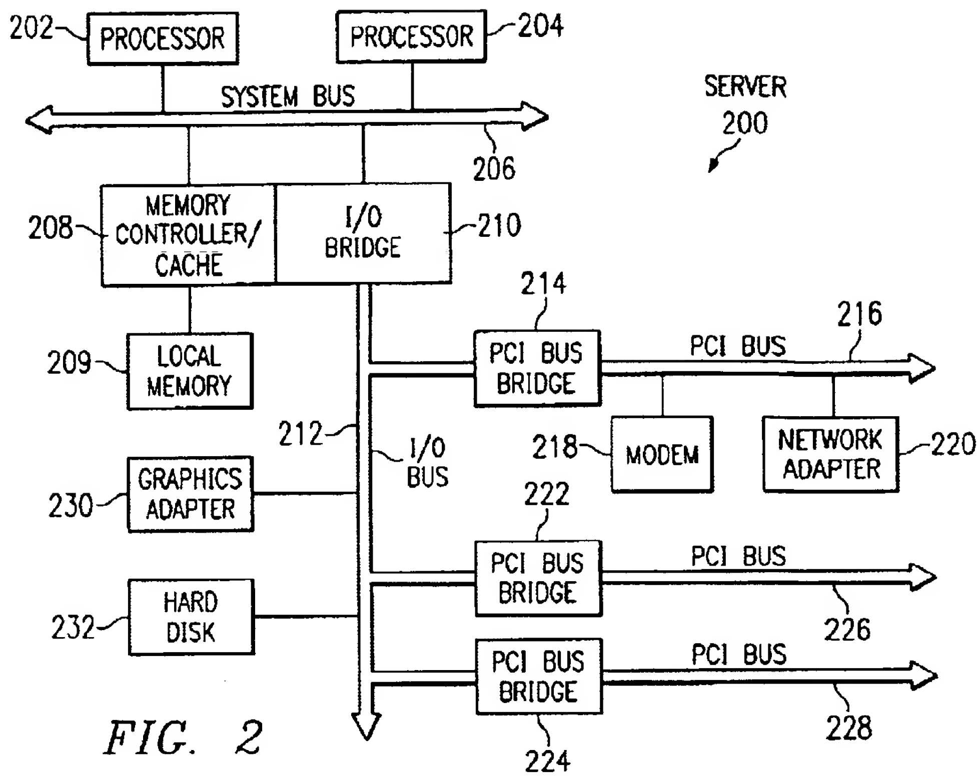

202
PROCESSOR
PROCESSOR
204
SYSTEM BUS
SERVER
200
206
208
MEMORY CONTROLLER/ CACHE
I/O BRIDGE
210
214
216
209
LOCAL MEMORY
PCI BUS BRIDGE
PCI BUS
212
I/O BUS
218
MODEM
NETWORK ADAPTER
220
230
GRAPHICS ADAPTER
222
PCI BUS BRIDGE
PCI BUS
226
232
HARD DISK
PCI BUS BRIDGE
PCI BUS
228
224

FIG. 2

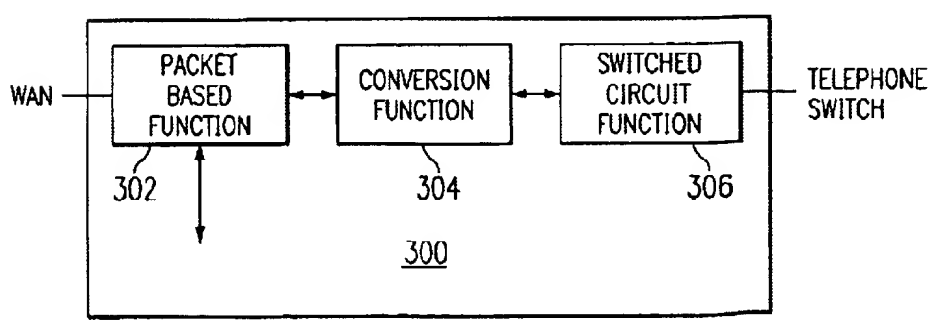

WAN
PACKET BASED FUNCTION
CONVERSION FUNCTION
SWITCHED CIRCUIT FUNCTION
TELEPHONE SWITCH
302
304
306
300

FIG. 3

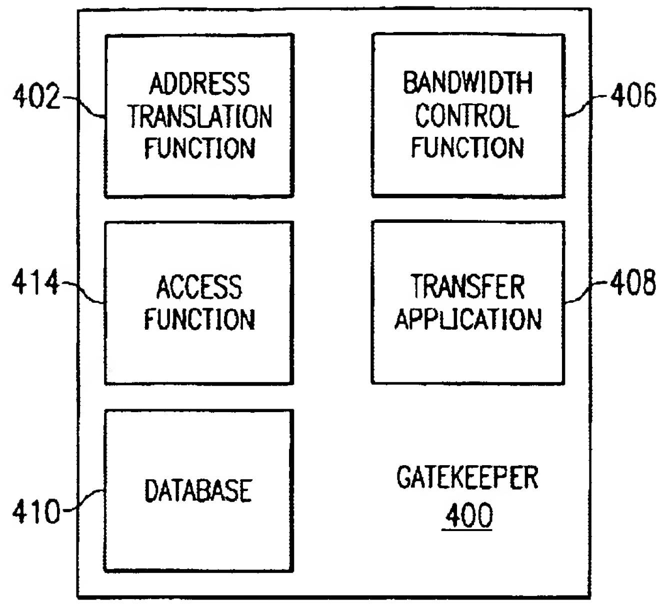


FIG. 4

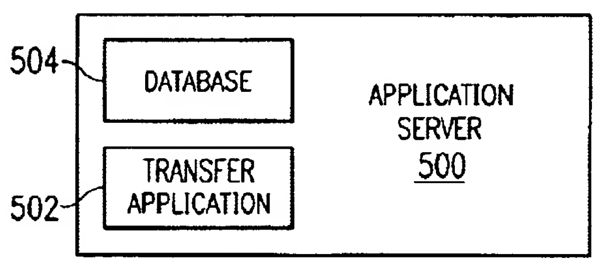


FIG. 5

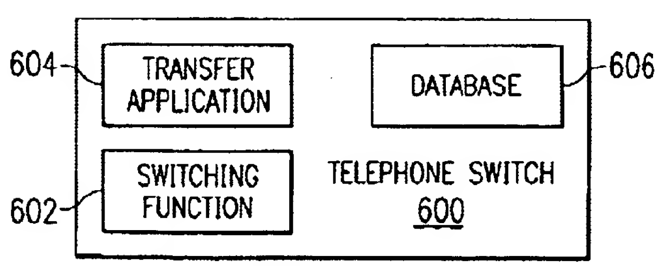
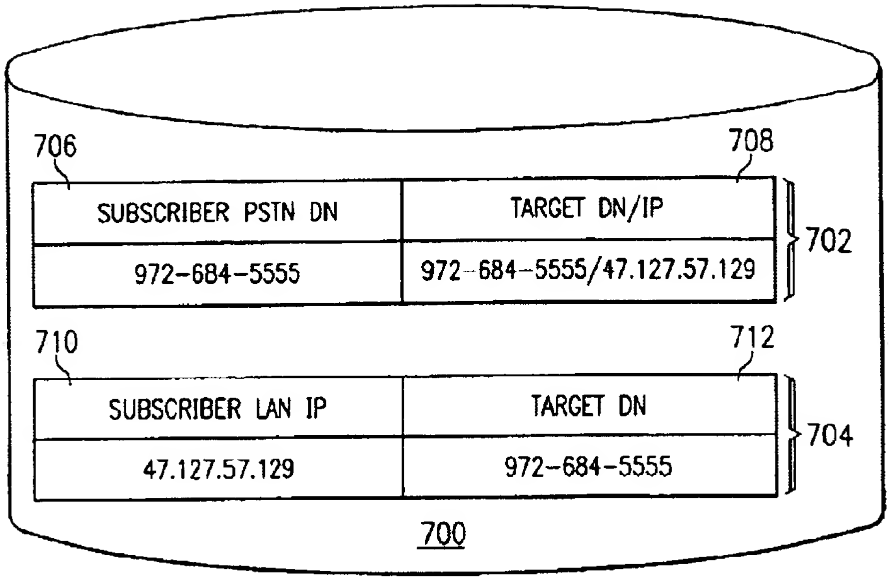


FIG. 6

FIG. 7



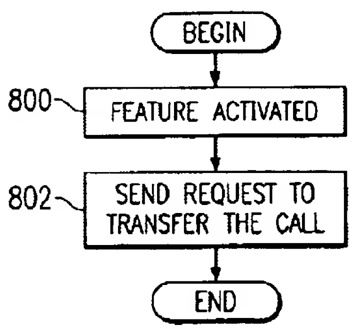


FIG. 8

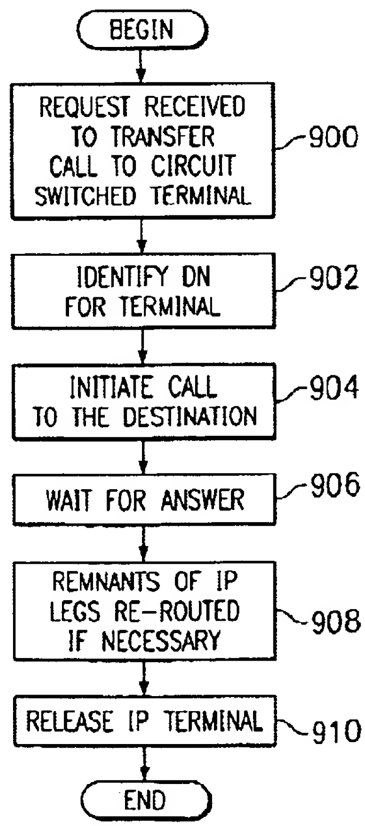


FIG. 9

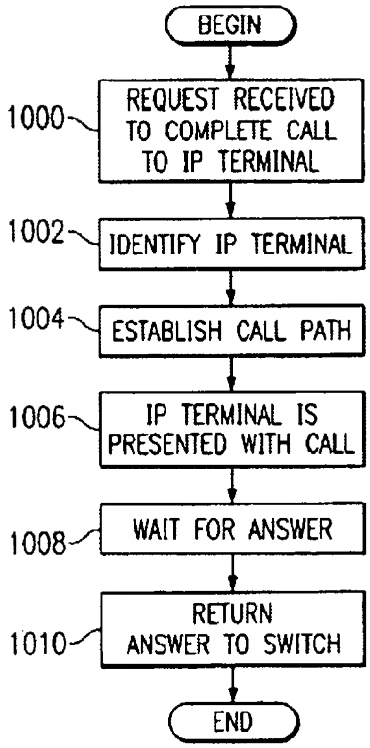


FIG. 10

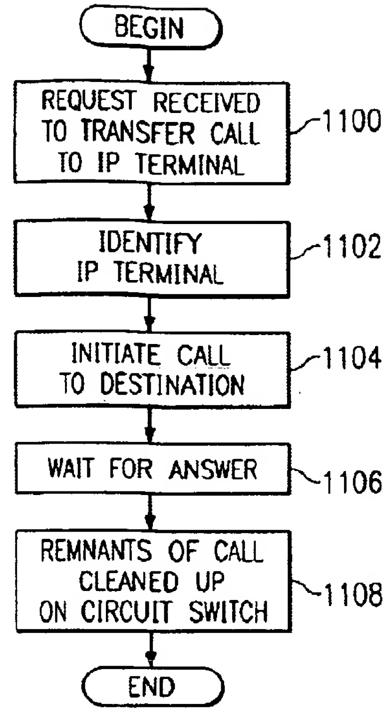


FIG. 11

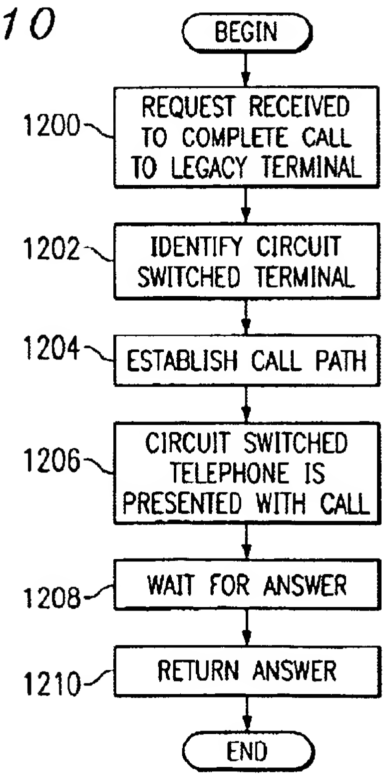


FIG. 12

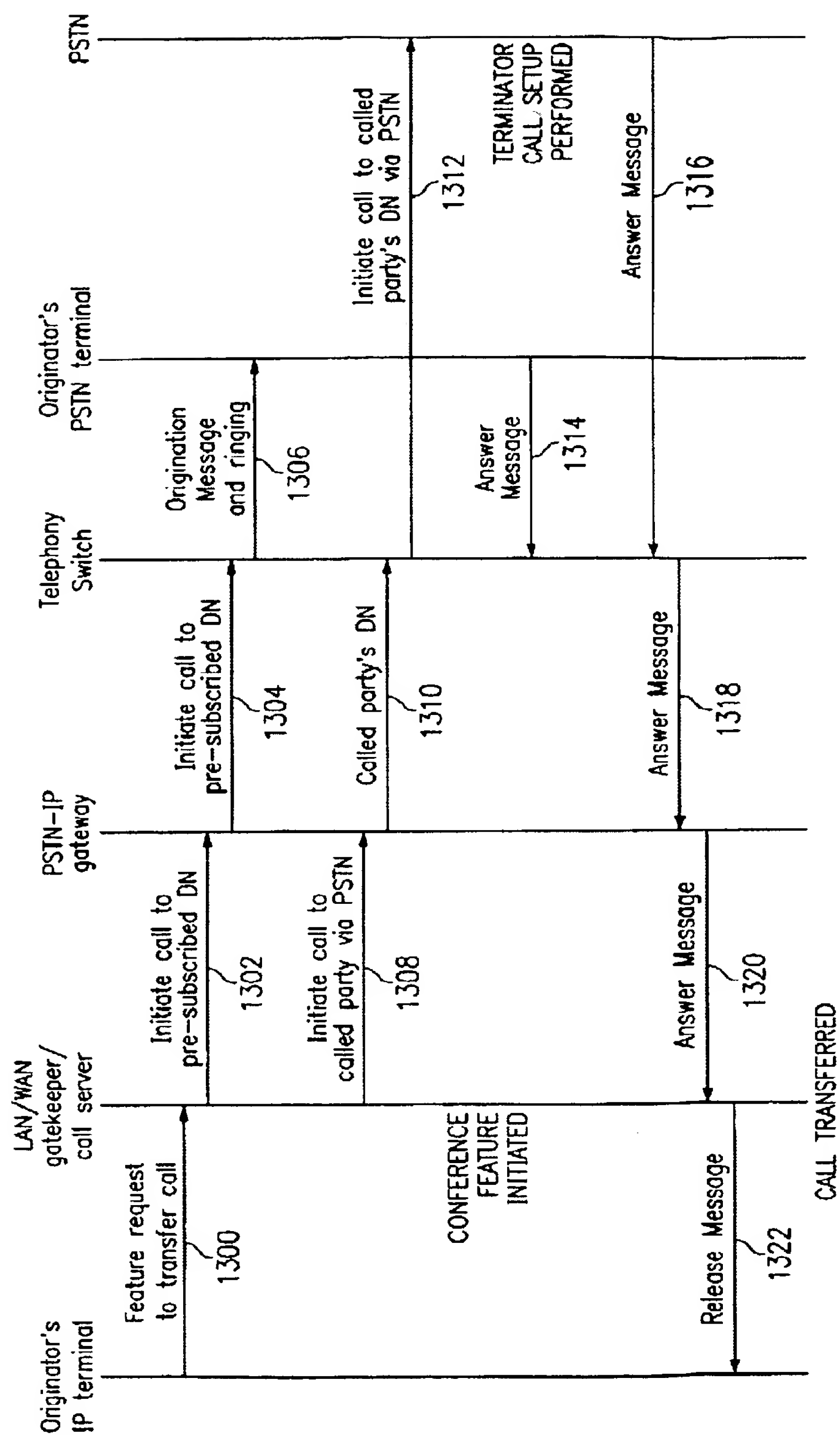

Originator's IP terminal
LAN/WAN gatekeeper/ call server
PSTN-IP gateway
Telephony Switch
Originator's PSTN terminal
PSTN
Feature request to transfer call
1300
Initiate call to pre-subscribed DN
1302
Initiate call to pre-subscribed DN
1304
Origination Message and ringing
1306
Initiate call to called party via PSTN
1308
Called party's DN
1310
Initiate call to called party's DN via PSTN
1312
TERMINATOR CALL SETUP PERFORMED
CONFERENCE FEATURE INITIATED
Answer Message
1314
Answer Message
1316
Answer Message
1318
Answer Message
1320
Release Message
1322
CALL TRANSFERRED

FIG. 13

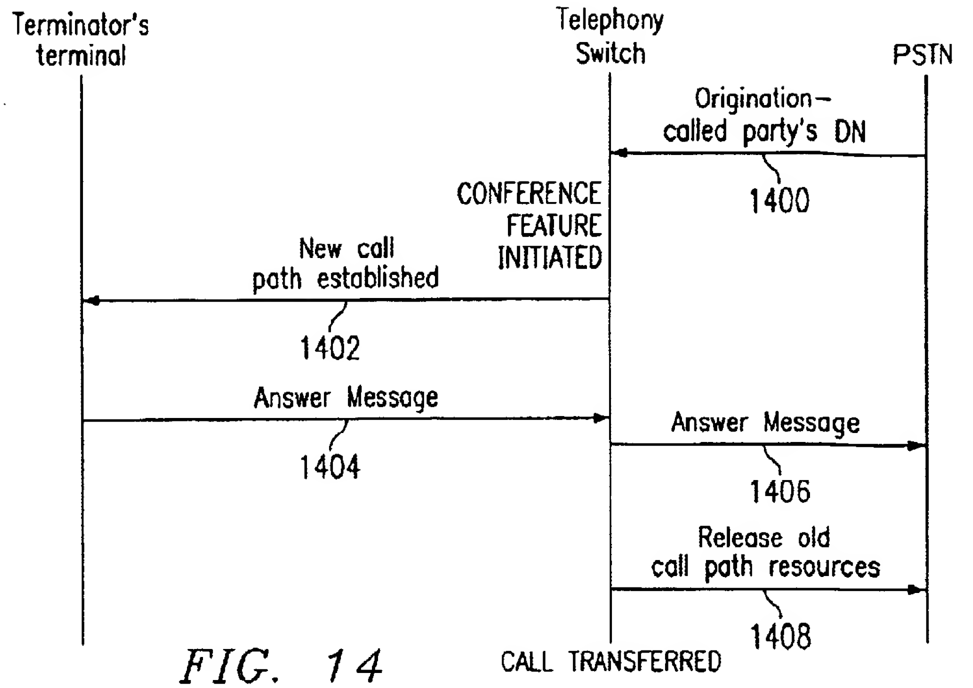

Terminator's terminal
Telephony Switch
PSTN
Origination– called party's DN
1400
CONFERENCE FEATURE INITIATED
New call path established
1402
Answer Message
1404
Answer Message
1406
Release old call path resources
1408
CALL TRANSFERRED

FIG. 14

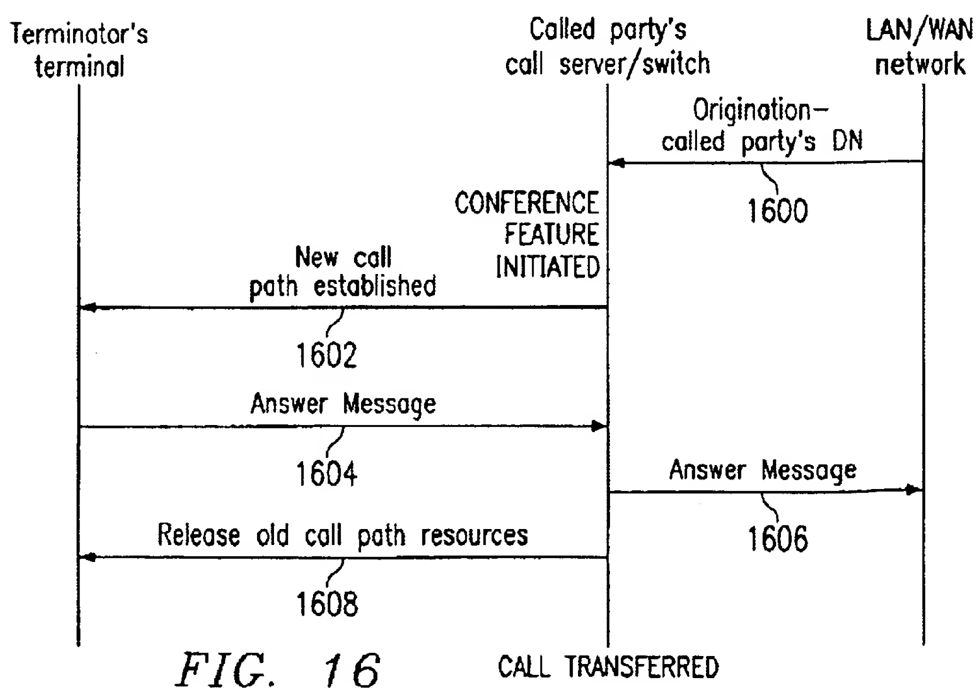

Terminator's terminal
Called party's call server/switch
LAN/WAN network
Origination– called party's DN
1600
CONFERENCE FEATURE INITIATED
New call path established
1602
Answer Message
1604
Answer Message
1606
Release old call path resources
1608
CALL TRANSFERRED

FIG. 16

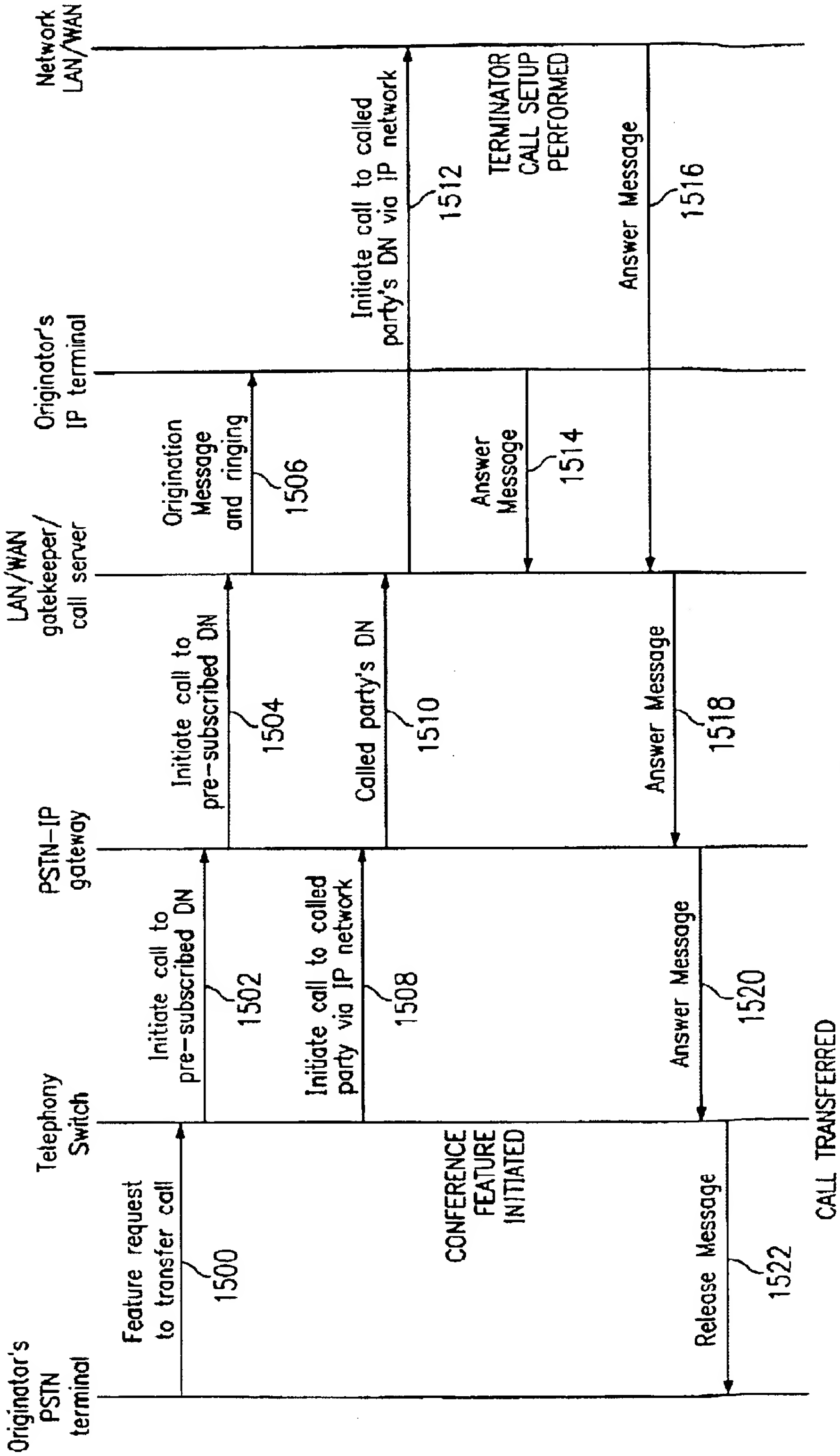

Originator's PSTN terminal
Telephony Switch
PSTN-IP gateway
LAN/WAN gatekeeper/ call server
Originator's IP terminal
Network LAN/WAN
Feature request to transfer call
1500
Initiate call to pre-subscribed DN
1502
Initiate call to pre-subscribed DN
1504
Origination Message and ringing
1506
CONFERENCE FEATURE INITIATED
Initiate call to called party via IP network
1508
Called party's DN
1510
Initiate call to called party's DN via IP network
1512
TERMINATOR CALL SETUP PERFORMED
Answer Message
1514
Answer Message
1516
Answer Message
1518
Answer Message
1520
Release Message
1522
CALL TRANSFERRED

FIG. 15

METHOD AND APPARATUS FOR VOICE OVER INTERNET PROTOCOL SWAPPING IN A COMMUNICATIONS SYSTEM

CROSS REFERENCE TO RELATED APPLICATIONS

The present invention is related to application entitled METHOD AND APPARATUS FOR AUTOMATIC TRANSFER OF A CALL IN A COMMUNICATIONS SYSTEM IN RESPONSE TO CHANGES IN QUALITY OF SERVICE, Ser. No. 09/358,994, filed even date hereof, and assigned to the same assignee.

1. Field of the Invention

The present invention relates generally to communications system and in particular to a method and apparatus for routing calls in a communications system. Still more particularly, the present invention relates to a method and apparatus for routing voice over Internet protocol calls within a communications system.

2. Background of the Invention

Originally regarded as a novelty, Internet telephony is attracting more and more users because it offers tremendous cost savings relative to the traditional public switch network (PSTN) users can bypass long distance carriers and their permanent usage rates and run voice traffic over the Internet for a flat monthly Internet access fee. Internet telephony involves the use of voice over Internet protocol also referred to as "voice over IP" or "VoIP". This protocol is packet based in contrast to the switch circuit system in a PSTN.

For example, user A in Austin wants to make a point-to-point phone call to user B in the company's London office. User A picks up the phone and dials an extension to connect with the gateway server, which is equipped with a telephony board and compression-conversion software ;the server configures the PBX to digitize the upcoming call. User A then dials the number of the London office, and the gateway server transmits the (digitized, IP-packetized) call over the IP-based wide area network (WAN) to the gateway at the London end. The London gateway converts the digital signal back to analog format and delivers it to the called party. With this calling system, expensive international long distance charges are virtually eliminated because the call is set up as a local call.

Users of communications system are increasingly mobile and require reliability in calls, such as, business calls. For example, user A may want to continue the conversation with the called party in London, but has to leave for an appointment. In such a situation, user A must terminate or hang up the voice over IP call and reinitiate the call on user A's mobile phone by redialing the called party's number. In another example, user A calls a party on a mobile phone while in transit to work. When user A reaches work, user A must hang up the call and redial the called party's number to start a new call, using voice over IP. In this manner, user A reduces costs for the call.

In addition, the level of reliability and sound quality expected by users is not always available with voice over IP calls. This situation is primarily caused by bandwidth limitations that lead to packet loss in the network. When congestion occurs, delays in packet transmission may occur, resulting in packets being lost or discarded. This packet loss causes gaps or periods of silence in the conversation between users. These gaps or periods of silence lead up to a "clipped-speech" effect. Such a situation is unsatisfactory

for most users and is unacceptable in business communications. As a result, when a user is dissatisfied with the quality of a voice over IP call, the user must hang up the call and redial the called party's number to initiate a new call using a legacy phone to continue the conversation with the called party.

Terminating and reinitiating calls in this manner is inconvenient for a caller. As a result, a caller may often times continue a call using a legacy phone, such as a mobile phone, rather than hanging up the legacy phone and redialing the called party's number on a terminal using voice over IP. Therefore, it would be advantageous to have an improved method and apparatus for allowing a user to take advantage of voice over IP without the user having to terminate a call in progress and redial a called party's number to initiate a new call to continue the conversation.

SUMMARY OF THE INVENTION

The inconveniences to a user desiring flexibility and mobility in a communications system providing calls over a packet based network, such as voice over IP, are minimized through the method and apparatus of the present invention. A request is received from a user, at a first terminal in a communications system, during a call to switch the call from a packet based network to a circuit switched network. The call is switched to a second terminal associated with the user in response to receiving the request. The second terminal uses the circuit switched network and the call is switched to the second terminal without terminating the call.

The present invention also provides for switching from a path in a circuit switched network to a packet based network in response to a request from a user. When a request is received from a terminal during the call, the call is switched to another terminal using the packet based network.

The switching of the call between a packet based network and a circuit switched network may be accomplished by establishing a new path to a new terminal on the desired network while the path through the present network continues to be used for the call. When the new path is established, the new path is joined to the call. The portion of the current path through the current network is released or discontinued. The joining of the paths may be accomplished through a call conferencing feature used to provide call conference functions. The destination for the call may be selected by associating the user with a preselected destination stored in a database, which is queried when the user makes a request to switch or transfer the call.

Other aspects and features of the present invention will become apparent to those ordinarily skilled in the art upon review of the following description of specific embodiments of the invention in conjunction with the accompanying figures.

BRIEF DESCRIPTION OF THE DRAWINGS

The novel features believed characteristic of the invention are set forth in the appended claims. The invention itself, however, as well as a preferred mode of use, further objectives and advantages thereof, will best be understood by reference to the following detailed description of an illustrative embodiment when read in conjunction with the accompanying drawings, wherein:

FIG. 1 is an illustration of a communications system depicted in accordance with a preferred embodiment of the present invention;

FIG. 2 is a block diagram depicting a data processing system that may be implemented as a server in accordance with a preferred embodiment of the present invention;

FIG. 3 is a block diagram of a gateway depicted in accordance with a preferred embodiment of the present invention;

FIG. 4 is a block diagram of a gatekeeper depicted in accordance with a preferred embodiment of the present invention;

FIG. 5 is a block diagram illustrating functions in an application server depicted in accordance with a preferred embodiment of the present invention;

FIG. 6 is a block diagram of functions in a switch depicted in accordance with a preferred embodiment of the present invention;

FIG. 7 is a diagram illustrating a database used in transferring calls depicted in accordance with a preferred embodiment of the present invention;

FIG. 8 is a flowchart of a process used at a terminal to transfer a call depicted in accordance with a preferred embodiment of the present invention;

FIG. 9 is a flowchart of a process for use in a transfer application located in a server or gatekeeper to transfer a call to a circuit switched network depicted in accordance with a preferred embodiment of the present invention;

FIG. 10 is a flowchart of a process for use in a transfer application located in a server or gatekeeper to transfer a call to a packet based network depicted in accordance with a preferred embodiment of the present invention;

FIG. 11 is a flowchart of a process for use in a transfer application located in a switch to transfer a call to a packet based network depicted in accordance with a preferred embodiment of the present invention;

FIG. 12 is a flowchart of a process for use in a transfer application located in a switch to transfer a call to a circuit switched network depicted in accordance with a preferred embodiment of the present invention;

FIG. 13 is a message flow diagram of a process used to transfer a call from an IP network to a PSTN depicted in accordance with a preferred embodiment of the present invention;

FIG. 14 is a message flow diagram of a process for transferring a call from an IP network to a PSTN depicted in accordance with a preferred embodiment of the present invention;

FIG. 15 is a message flow diagram of a process for transferring a call from a PSTN to an IP network depicted in accordance with a preferred embodiment of the present invention; and

FIG. 16 is a message flow diagram of a process used to transfer a call from a PSTN to an IP network depicted in accordance with a preferred embodiment of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

With reference now to the figures and in particular with reference to FIG. 1, an illustration of a communications system is depicted in accordance with a preferred embodiment of the present invention. Communications system 100 includes a public switch telephone network (PSTN) 102, a wide area network (WAN) 104, and a local area network (LAN) 106. Telephone switch 108 and wireless switch 110 are part of PSTN 102. PSTN 102 is a circuit switched network while WAN 104 and LAN 106 are packet-based networks. Communications system 100 also includes gateways 112–116. Gateway 112 provides an interface between WAN 104 and wireless switch 110. Gateway 114 provides an interface between WAN 104 and telephone switch 108. Gateway 116 provides an interface between switch (not shown) in PSTN 102. A gatekeeper 118 and an application server 120 are connected to WAN 104. Gateways 112–116 provide translation of calls into the appropriate protocols for use in WAN 104 and PSTN 102. Gatekeeper 118 provides network management functions in WAN 104. Application server 120 provides call functions for voice over IP calls as well as for other types of applications on WAN 104.

Terminals in the form of a personal computer (PC) 122 and an IP terminal 124 are located in location 126. In the depicted examples, location 126 may be a residence or office. PC 122 is connected to LAN 106. IP terminal 124 also is connected to LAN 106. IP terminal 124 may be, for example, a telephone configured for communication over a packet-based network (e.g. WAN 104).

Location 128 in communications system 100 includes PC 130, cordless phone 132, and telephone 134. These terminals are connected to telephone switch 108. Cordless phone 132 and telephone 134 are legacy phones. A legacy phone in the depicted examples is a conventional landline or circuit switched telephone for use with PSTN 102. Also found in communications system 100 are mobile stations 136–140. These mobile stations are legacy mobile phones, which are circuit switched terminals that communicate using switch circuit networks.

Communications system 100 as depicted in FIG. 1 is intended as an illustrative example of a communications system in which the present invention may be implemented and not as an architectural limitation. For example, WAN 104 may be placed by the Internet, which is a worldwide collection of networks. Further, Intranets also may be present within communications system 100. Of course, other elements not shown may be contained within communications system 100.

The present invention provides a method and apparatus for use in a communications system, such as communications system 100 in FIG. 1, to conveniently route calls for a user wishing to switch terminals. Specifically, the present invention allows the user to move or transfer an existing voice call back and forth from a packet based network (e.g. voice over IP network or IP network) and a switch circuit network (e.g. a legacy telephone switching system).

For example, a user may initiate a voice over IP call from a terminal, such as PC 122 at location 126. The call may be to a terminal such as, PC 130, cordless telephone 132, or telephone 134 at location 128 travels though a packet based network, LAN 106 and WAN 104 before routed through a circuit switched network, PSTN 102, to reach a terminal at location 128. If the user at PC 122 desires to leave location 126 and continue the call, the mechanism of the present invention allows the user to initiate a transfer of the call from PC 122 to another terminal, such as mobile station 136. In activating such a feature, the mechanism of the present invention may place the existing call on hold and transfer it to the appropriate mobile station. Upon answering the phone at mobile station 136, the mechanism of the present invention would allow the call to continue at mobile station 136 without requiring the user to reinitiate the call.

From a mobile station or other type of legacy telephone, a user may transfer a call from the legacy telephone to a voice over IP capable terminal to continue the call without having to redial the called party's phone number. For example, a caller on mobile station 138 initiates a call to telephone 134 at location 128 while traveling to location

126. When the user reaches location 126, the user may activate the mechanism of the present invention during a call and have the call transferred to IP terminal 124 at location 126. This transfer occurs without the user having to redial or reinitiate the call to the called party at telephone 134. By activating the mechanism of the present invention, the call is put on hold while IP terminal 124 is called. When the user answers the call at terminal 124 a transfer of the call completes.

Activation of the mechanism of the present invention occurs while the call is in progress and may be activated by a number of different ways. For example, on a legacy telephone, the mechanism may be activated by using a feature key and dialing a feature access code on a phone. On a PC, the feature may be activated by selection of an appropriate function key or icon on a graphic user interface.

Referring to FIG. 2, a block diagram depicts a data processing system that may be implemented as a server in accordance with a preferred embodiment of the present invention. Data processing system 200 may be implemented as application server 120 in FIG. 1 or as a database server, such as a quality of service database server 118. Data processing system 200 may be a symmetric multiprocessor (SMP) system including a plurality of processors 202 and 204 connected to system bus 206. Alternatively, a single processor system may be employed. Also connected to system bus 206 is memory controller/cache 208, which provides an interface to local memory 209. I/O bus bridge 210 is connected to system bus 206 and provides an interface to I/O bus 212. Memory controller/cache 208 and I/O bus bridge 210 may be integrated as depicted.

Peripheral component interconnect (PCI) bus bridge 214 connected to I/O bus 212 provides an interface to PCI local bus 216. A number of modems may be connected to PCI bus 216. Typical PCI bus implementations will support four PCI expansion slots or add-in connectors. Communications links to network computers 108–112 in FIG. 1 may be provided through modem 218 and network adapter 220 connected to PCI local bus 216 through add-in boards.

Additional PCI bus bridges 222 and 224 provide interfaces for additional PCI buses 226 and 228, from which additional modems or network adapters may be supported. In this manner, server 200 allows connections to multiple network computers. A memory-mapped graphics adapter 230 and hard disk 232 may also be connected to I/O bus 212 as depicted either directly or indirectly.

Those of ordinary skill in the art will appreciate that the hardware depicted in FIG. 2 may vary. For example, other peripheral devices, such as optical disk drives and the like also may be used in addition to or in place of the hardware depicted. The depicted example is not meant to imply architectural limitations with respect to the present invention.

The data processing system depicted in FIG. 2 may be, for example, an IBM RISC/System 6000 system, a product of International Business Machines Corporation in Armonk, N.Y., running the Advanced Interactive Executive (AIX) operating system. Furthermore, data processing system 200 may be implemented as an end user PC, such as end user PC 122 or end user PC 130. An adapter allowing a user to place voice calls would be added for use in an end user PC.

FIGS. 3–6 are block diagrams illustrating examples of components that may be used to implement the processes of the present invention. With reference now to FIG. 3, a block diagram of a gateway is depicted in accordance with a preferred embodiment of the present invention. Gateway 300 is an example of a gateway, such as gateways 112, 114, or 116 in FIG. 1. Gateway 300 provides all of the logical and electrical translation functions required to provide communications between a packet based environment, such as WAN 104, and a circuit switched environment, such as PSTN telephone switch 110 in FIG. 1. The functions of gateway 300 are implemented along International Telecommunications Union's ITU-T Recommendation H.323, which is a standard describing systems and equipment for use in providing multi-media communications over packet based networks. Gateway 300 in FIG. 3 contains a packet-based function 302, which serves to communicate with packet based network devices. Switch circuit function 306 in gateway 300 is employed to communicate with circuit switched network devices. Conversion function 304 provides protocol conversion functions for conversion of data and other signals sent between the two environments. For example, if a voice call is flowing from a telephone switch to a WAN using gateway 300, the telephony-based traffic is compressed and placed into IP packets and routed on to the WAN.

With reference now to FIG. 4, a block diagram of a gatekeeper is depicted in accordance with preferred embodiment of the present invention. Gatekeeper 400 provides network management functions. Gatekeeper 400 provides call control in routing, basic telephony services, bandwidth allocation, total network usage control, and system administration and security policies. Gatekeeper 400 contains address translation function 402, which provides address translation on a packet-based network. For example, address translation function 402 may provide a directory service allowing users to enter an alias address, which is converted into a network address for the terminal to use. The alias address may be, for example, a telephone number, an extension number, or a name. Access function 414 provides the access control used to determine whether a terminal can make or accept a call. Access function 414 is used to determine whether a terminal is allowed to access gateways to make outside calls. Bandwidth function 406 is used to determine the amount of bandwidth that is allocated for a call.

Gatekeeper 400 also includes a transfer application 408, which is used to provide the transfer functions of the present invention. In particular transfer application 408 will receive signals or requests from a user to transfer a voice over IP call using a packet-based network to a legacy telephone system call using a circuit switched network. The destination for the transferred call and information used to transfer the call is obtained from database 410 in the depicted examples.

With reference now to FIG. 5, a block diagram illustrating functions in an application server is depicted in accordance with a preferred embodiment of the present invention. Application server 500 is an example of an application server, such as application server 120 in FIG. 1. In the depicted examples, application server 500 may include a transfer application 502 similar to transfer application 408 located within gatekeeper 400 in FIG. 4. Transfer application 502 will detect requests to transfer calls between packet-based networks and circuit switched networks. The information for identifying the destination and transferring the call is located in database 504 in applications server 500.

Turning to FIG. 6, a block diagram of functions in a switch is depicted in accordance with a preferred embodiment of the present invention. Switch 600 may be implemented as telephone switch 108 or wireless switch 110 in FIG. 1. Switch 600 includes switching function 602, which is used to route calls. Additionally, transfer application 604

is present within switch 600. This application will transfer a call to a path through a circuit switched network in response to a signal or call from a user at a terminal to transfer the call. The information used to transfer the call is found in database 606.

With reference now to FIG. 7, a diagram illustrating a database used in transferring calls is depicted in accordance with a preferred embodiment of the present invention. Database 700 contains entries, such as entries 702 and 704, which contain information used to transfer calls between a packet based network and a circuit switched network.

Entry 702 contains an example of information used by a switch to transfer a call. Entry 702 is found in a database used by a switch, such as database 606 in FIG. 6. Entry 702 includes a subscriber PSTN directory number (DN) field 706 and a target DN/IP field 708. Subscriber PSTN DN fields 706 contains the subscribers telephone number. In this example, the number is "972-684-5555". This field is used to identify the target destination when a request is received from a user to transfer a call. The target destination is located in target DN/IP field 708. This field includes a telephone number and/or IP address of the destination. In this example, the target DN/IP is "972-684-5555/47.127.157.129". The directory number may be used by some networks to reach a terminal. In such a network, a gatekeeper provides a directory service to translate the directory number into the appropriate IP address. The IP address is the address of the terminal on the network. Other address formats may be used depending on the protocol employed. This information is used by a transfer application to forward or transfer a call to the target destination from the terminal at which the user requested the transfer.

Next, entry 704 is an example of an entry in a database found in a gatekeeper, such as database 410 in FIG. 4. Entry 704 includes a subscriber LAN IP field 710, which contains an identification of the terminal at which the user is located. The entry in this field is an IP address in the depicted examples, but may be another type of address depending on the type of protocol used on the network. In this example, the address is "47.127.57.129". Target DN field 712 contains the directory number of the target destination to which the call must be transferred if the user activates the feature. In this example, the target DN is "972-684-5555" and is directed towards a destination and a circuit switched network. Entry 704 also may be used in database 504 in application server 500 in FIG. 5.

With reference now to FIG. 8, a flowchart of a process used at a terminal to transfer a call is depicted in accordance with a preferred embodiment of the present invention. A user may desire to transfer a call from a packet-based network to a circuit switched network for various reasons. For example, if the quality of service on a packet-based network is unacceptable or the user wants more mobility, the user will use this process to transfer the call. A user may transfer a call from a circuit switched to a packet-based network for various reasons. For example, the user may desire to obtain cheaper rates for the call. Also, the user may desire to use the terminal on the packet-based network to send a data file to the called party while continuing the conversation with the call party.

The process begins by activating the feature (step 800). A user of a terminal may activate the feature in a number of ways. For example, if a user on an IP terminal wants to transfer an existing voice over IP telephone call to a traditional legacy terminal, such as a telephone, the user would depress a feature key or enter an access code to activate the transfer feature. On a legacy terminal, such as a mobile phone, a landline phone, or a cordless phone, the same type of mechanism may be used to activate the transfer feature of the present invention. Thereafter, a request is sent by the terminal to the switch to transfer the call (step 802). The request is sent to a switch, a server, or a gatekeeper in the depicted examples. A request transferring the call from a path using a packet based network to a path using a switch circuit network is sent to a switch. The request is sent to a server or gatekeeper if the requested transfer is from a circuit switched network to a packet based network.

Optionally, the user could enter a directory number or IP address for the desired destination. Such a selection of the destination may be used in place of a preselected destination for the user stored in a database.

Turning next to FIG. 9, a flowchart of a process for use in a transfer application located in a server or gatekeeper to transfer a call to a circuit switched network is depicted in accordance with a preferred embodiment of the present invention. This process is employed to transfer a call on a packet-based network to a circuit switched network.

The process begins by receiving a request to transfer the call to a circuit switched terminal on a circuit switched network (step 900). The directory number for the terminal for the user is identified by querying a database of subscribers located at the server (step 902). The IP address of the terminal originating the request for the transfer is mapped to a predefined directory number, which is accessible via the circuit switched network. Specifically, the IP address is used to locate the entry for the user. This entry is an entry, such as, for example, entry 704 in FIG. 7. The server then initiates a call to the destination selected for the user (step 904). The process then waits for an answer from the terminal at the destination (step 906). The process then reroutes any remnants of the path if needed (step 908). The process then releases the IP terminal (step 910) with the process terminating thereafter.

With reference now to FIG. 10, a flowchart of a process for use in a transfer application located in a server or gatekeeper to transfer a call to a packet based network is depicted in accordance with a preferred embodiment of the present invention. This process is used to handle a request from a switch to transfer a call to a packet-based network. The process begins by receiving a request to complete a call to an IP device (step 1000). The IP terminal to which the call is to be sent is identified (step 1002). This step is accomplished by taking the directory number from the request and mapping it to a pre-identified IP address to which the directory number is associated. The directory number may be used to identify an entry, such as entry 704 in FIG. 7. In the depicted example, the directory number is the directory number at which the user is located. Alternatively, the request could include the IP address of the user or use directory functions in the gatekeeper. Upon locating the address for the terminal, a call path is established to the terminal (step 1004). Thereafter, the IP terminal is presented with the call (step 1006). The server then waits for an answer (step 1008). When an answer occurs at the IP terminal, the answer is returned to the switch (step 1010) with the process terminating thereafter.

With reference next to FIG. 11, a flowchart of a process for use in a transfer application located in a switch to transfer a call to a packet based network is depicted in accordance with a preferred embodiment of the present invention. This process is employed in a transfer application, such as transfer application 604 in FIG. 6, to move or transfer an

existing call from a legacy terminal on a circuit switched network to an IP terminal on a packet based network.

The process beings by receiving a request to transfer the call to an IP terminal (step **1100**). The request is received from a user on a legacy terminal. Next, IP terminal for the user is identified by querying a database of subscribers located at the switch (step **1102**). This query locates an entry, such as entry **702** in FIG. **7**. Entry **702** provides a target DN or IP address for the terminal on the packet-based network to which the call is to be forwarded. Thereafter, the switch initiates a call to the destination identified in the entry (step **1104**). The switch then waits for an answer (step **1106**). When an answer is received, the switch will clean up remnants of the call (step **1108**) with the process terminating thereafter. Since the path for the transferred call is optimized to not pass through the switch if the switch is not required, the switch releases all the hardware and software resources previously associated with the call. These resources include, for example, trunk circuits, PSTN telephones, and time division multiplex (TDM) network connections and/or paths.

With reference now to FIG. **12**, a flowchart of a process for use in a transfer application located in a switch to transfer a call to a circuit switched network is depicted in accordance with a preferred embodiment of the present invention. This process is initiated in response to a request from a server in a packet-based network to transfer a call to a legacy terminal in a circuit switched network.

The process begins by receiving a request to complete a call to a circuit switched terminal (step **1200**). The circuit switched terminal to which the call is to be sent is identified (step **1202**). This step is accomplished by taking the IP address from the request and mapping it to a pre-identified directory number to which the IP address is associated. Upon locating the directory number for the terminal, a call path is established to the terminal (step **1204**). Thereafter, the circuit switched terminal is presented with the call (step **1206**). The switch then waits for an answer (step **1208**). When an answer occurs at the circuit switched terminal, the answer is returned to the server (step **1210**) with the process terminating thereafter.

The message flow diagrams described below in FIGS. **13–16** illustrate examples of the present invention. The diagrams are described with respect to a packet-based network in the form of an IP network and with respect to a circuit switched network in the form of a PSTN using time division multiplexing (TDM). These flows can be applied to other types of networks other than the ones described. For example, without limitation, packet based networks, such a frame relay and asynchronous transfer mode (ATM) networks also may used.

Turning now to FIG. **13**, a message flow diagram of a process used to transfer a call from an IP network to a PSTN is depicted in accordance with a preferred embodiment of the present invention. This message flow diagram illustrates the sequence of messages occurring when a user originating a call requests the call to be transferred. A request is received by a gatekeeper or other call server from an IP terminal on an IP network to transfer the call from the IP network to the PSTN (step **1300**). This terminal is the originator's IP terminal. In response, the gatekeeper sends a message to a gateway to initiate a call to a pre-subscribed directory number (step **1302**). The gateway forwards this message to a telephone switch, which is also referred to as a "first switch" in these examples (step **1304**). The switch uses the directory number to send an origination message to the PSTN terminal identified using the directory number and to cause the terminal to ring (step **1306**). The gatekeeper also sends a request to the gateway to initiate a call to the called party's directory number (step **1308**). The gateway sends the called party's called directory number to a second switch (step **1310**). This switch initiates a call to the called party's directory number through the PSTN (step **1312**). An answer message is from originator's terminal on the PSTN (step **1314**). This occurs when the user picks up or answers the PSTN telephone. An answer is received from the PSTN when a new path is established by the second switch to the called party's terminal (step **1316**). Steps **1312–1316** involve the use of existing conference features to put the calls together. The steps involve the use of a conference bridge located in the second switch. The answers are returned in a message from the switch to the gateway (step **1318**). The gateway relays the message to the gatekeeper (step **1320**). In response, the gatekeeper sends a release message to the IP terminal to release the IP terminal from the call (step **1322**).

In FIG. **14**, a message flow diagram of a process for transferring a call from an IP network to a PSTN is depicted in accordance with a preferred embodiment of the present invention. This flow is from the perspective of the second switch described in FIG. **13**.

A message is received by the switch from first switch in the PSTN with a party's directory number (step **1400**). A conference feature is initiated by the switch using a conference bridge in the switch. A new call is established to the terminator's terminal (step **1402**). At this time a path is present from the switch to the called parties terminal to first port in the conference bridge. A second path is present from the original IP based call from the originating IP terminal to a second port in the conference bridge. Another path is present that leads from a third port in the conference bridge to the originator's PSTN telephone. The conference port will put the paths together to "conference" the call. An answer is received by the switch from the called party's terminal (step **1404**). The called party need not take any action in this case. The answer is an acknowledgement that the path is present. The party will hang up or terminate the call from the IP terminal or the gatekeeper can drop the call to the IP terminal. At that time, the switch will release the port to the IP terminal and the call on the other two ports will continue.

Thereafter, the switch sends the answer message to the first switch in the PSTN (step **1406**). The switch also releases the resources in the old call path by sending a request to the first switch in the PSTN (step **1408**). At this time the call has been successfully transferred.

With reference to FIG. **15**, a message flow diagram of a process for transferring a call from a PSTN to an IP network is depicted in accordance with a preferred embodiment of the present invention. A request is received from an PSTN terminal by a switch to transfer the call to an IP network (step **1500**). In response, the switch sends a message to a gateway to initiate a call to a pre-subscribed directory number (step **1502**). The gateway sends the request to a gatekeeper or other application server on the IP network (step **1504**). Thereafter, an origination message is sent to the originator's IP terminal and the IP terminal rings or indicates a call is present to be answered (step **1506**). The switch also sends a request to the gateway to initiate a call to the called party through the IP network (step **1508**). The gateway also will send the called party's directory number from the request to the gatekeeper (step **1510**). The gatekeeper then initiates a call to the called party's directory number to the IP network (step **1512**). An answer is received from the

originator's IP terminal (step **1514**). An answer also is received from the packet-based network (step **1516**). Steps **1512–1516** are steps used in conference call functions to join calls as described above.

An answer message is received by the server to the gateway in response to the gatekeeper receiving the answers from the two terminals (step **1518**). The message is relayed by the gateway to the switch (step **1520**). In turn, the switch will send a release message to the originator's PSTN terminal (step **1522**).

Next, FIG. **16** is a message flow diagram of a process used to transfer a call from a PSTN to an IP network is depicted in accordance with a preferred embodiment of the present invention. A message is received by a switch from the WAN in the IP network with a called parties directory number (step **1600**). A conference call feature is initiated by the switch. A new call path is established by the switch to the terminator's terminal (step **1602**). An answer message is returned by the terminator's terminal to the switch (step **1604**). The answer message is sent by the switch to the network (step **1606**). In response, the switch will send a message to release old call path resources (step **1608**). At this point, the call has been transferred.

Thus, the present invention provides an improved method an apparatus for transferring calls without the user having to hang up or terminate a call at the terminal and reinitiate the call at a desired terminal. The present invention provides this advantage by allowing the call to be automatically transferred in response to an activation of a feature. The call is transferred to a pre-selected destination at which the user desires to continue the call. In this manner, a user may switch the path of a call in progress back and forth between a packet-based network and a circuit switched network.

It is important to note that while the present invention has been described in the context of a fully functioning data processing system, those of ordinary skill in the art will appreciate that the processes of the present invention are capable of being distributed in the form of a computer readable medium of instructions and a variety of forms and that the present invention applies equally regardless of the particular type of signal bearing media actually used to carry out the distribution. Examples of computer readable media include recordable-type media such a floppy disc, a hard disk drive, a RAM, and CD-ROMs and transmission-type media such as digital and analog communications links.

The description of the present invention has been presented for purposes of illustration and description, but is not intended to be exhaustive or limited to the invention in the form disclosed. Many modifications and variations will be apparent to those of ordinary skill in the art. The various illustrated components used in transferring calls may be placed in different locations in the communications system other than those in the depicted examples. For example, the transfer application is illustrated as being located in an application server or gatekeeper in a packet based network in the examples. The transfer application may be located in other places within a packet-based network, such as a call server. Some of the processes in the transfer application may be split out. For example, the directory translation may be performed at a terminal in which the terminal provides the directory number of the target. Although the depicted examples involve voice over IP, the processes may be applied to other packet-based protocols. Further, although the examples used a conferencing mechanism to make the transfer, other mechanism may be used. For example, the call between the parties could be put on hold such that the called party hears nothing or music on hold momentarily until the caller picks up at the terminal to which the call was transferred.

The embodiment described was chosen and described in order to best explain the principles of the invention, the practical application, and to enable others of ordinary skill in the art to understand the invention for various embodiments with various modifications as are suited to the particular use contemplated.

What is claimed is:

1. A method in a communications system for routing a call, the method comprising:

receiving a request from a user, at a first terminal, during a call to switch the call from a packet based network to a circuit switched network; and

responsive to receiving the request, switching the call to a second terminal associated with the user, wherein the second terminal uses the circuit switched network and wherein the call is switched to the second terminal without terminating the call;

wherein the step of switching comprises:

creating a path from the second terminal on the circuit switched network to a destination terminal; and

continuing the call using the path on the circuit switched network.

2. A method in a communications system for routing a call, the method comprising:

receiving a request from a user, at a first terminal, during a call to switch the call from a packet based network to a circuit switched network; and

responsive to receiving the request, switching the call to a second terminal associated with the user, wherein the second terminal uses the circuit switched network and wherein the call is switched to the second terminal without terminating the call;

establishing a path from the second terminal on the circuit switched network to the destination terminal; and

responsive to an answer at the second terminal, using the path to continue the call without interruption.

3. A method in a communications system for routing a call, the method comprising:

receiving a request from a user, at a first terminal, during a call to switch the call from a packet based network to a circuit switched network; and

responsive to receiving the request, switching the call to a second terminal associated with the user, wherein the second terminal uses the circuit switched network and wherein the call is switched to the second terminal without terminating the call;

wherein the first terminal has a first path to a switching node and a second path from the switching node to a destination node and wherein the step of switching comprises:

creating a third path to the second terminal on the circuit switched network; and

joining the third path to the second path, wherein the second terminal is joined into the call.

4. The method of claim **3** further comprising:

joining the third path to the first path, wherein the second terminal is joined into a conference call.

5. The method of claim **4**, wherein the first path, the second path, and the third path are joined at a conference bridge in the switching node.

6. A method in a communications system for routing a call, the method comprising:

receiving a request from a user, at a first terminal, during a call to switch the call from a packet based network to a circuit switched network; and

responsive to receiving the request, switching the call to a second terminal associated with the user, wherein the second terminal uses the circuit switched network and wherein the call is switched to the second terminal without terminating the call;

wherein the second terminal is associated with the user in a database; and

wherein the database is located in a switch.

7. A method in a communications system for routing a call, the method comprising:

receiving a request from a user, at a first terminal, during a call to switch the call from a circuit switched network to a packet based network; and

responsive to receiving the request, switching the call to a second terminal associated with the user, wherein the second terminal uses the packet based network and wherein the call is switched to the second terminal without terminating the call.

8. The method of claim 7, wherein the step of initiating comprises:

establishing a path to the second terminal on the packet based network; and

responsive to an answer at the second terminal, using the path to continue the call without interruption.

9. A communications system comprising:

a circuit switched network;

a packet based network;

a first terminal connected to the packet based network;

a second terminal connected to the circuit switched network;

a switch connected to the circuit switched network and the packet switched network, wherein the switch has a plurality of modes of operations including:

a first mode of operation, responsive to receiving a request to switch a request from the first terminal to switch to an active call between the first terminal and a destination terminal from the packet based network to the circuit switched network, in which the switch establishes a first path to the second terminal; and

a second mode of operation, responsive to establishing the first path, in which the switch joins the first path to a second path to the first terminal and to a third path to the destination terminal.

10. The communications system of claim 9, wherein the switch joins the first path to the second path and the third path using a conference bridge.

11. The communications system of claim 9, wherein the second terminal is a wireless telephone.

12. A switch comprising:

a plurality of communications ports;

a switch fabric connected to the plurality of communications ports; and

a processing unit, wherein the processing unit controls routing of calls through the switch fabric;

wherein the processing unit, responsive to receiving a request from a first terminal in communication with the switch through a first path using a packet based network to move a call in progress to a circuit switched network, establishes a second path to a second terminal associated with a user and joins the second path to the call in progress, wherein the second terminal is used to continue the call in progress; and

wherein the processing unit sends a message to release at least a portion of the first path using the packet based network such that the second path and any remaining portion of the first path use the circuit switched network.

13. A computer comprising:

a communications unit, wherein the communications units handles receiving and transmitting packets in a packet based network; and

a processing unit, wherein the processing unit routes packets received by the communications unit and wherein responsive to receiving a request from a terminal handling a voice call in progress to reroute the voice call in progress to a circuit switched network, the processing unit sends a request to a circuit switched network to reroute the call through the circuit switched network.

14. The computer of claim 13, wherein the processing unit routing releases the terminal in response to receiving an indication that a path has been established through the circuit switched network.

15. The computer of claim 13, wherein the indication is an answer message received from the circuit switched network.

16. The computer of claim 13, wherein the computer is gatekeeper.

17. The computer of claim 13, wherein the computer is a call server.

18. A communications system for routing a call, the communications system comprising:

receiving means for receiving a request from a user, at a first terminal, during a call to switch the call from a packet based network to a circuit switched network; and

switching means, responsive to receiving the request, for switching the call to a second terminal associated with the user, wherein the second terminal uses the circuit switched network and wherein the call is switched to the second terminal without terminating the call;

wherein the switching means comprises:

creating means for creating a path from the second terminal on the circuit switched network to a destination terminal; and

continuing means for continuing the call using the path on the circuit switched network.

19. A communications system for routing a call, the communications system comprising:

receiving means for receiving a request from a user, at a first terminal, during a call to switch the call from a packet based network to a circuit switched network; and

switching means, responsive to receiving the request, for switching the call to a second terminal associated with the user, wherein the second terminal uses the circuit switched network and wherein the call is switched to the second terminal without terminating the call;

establishing means for establishing a path from the second terminal on the circuit switched network to the destination terminal; and

using means, responsive to an answer at the second terminal, for using the path to continue the call without interruption.

20. A communications system for routing a call, the communications system comprising:

receiving means for receiving a request from a user, at a first terminal, during a call to switch the call from a packet based network to a circuit switched network; and

switching means, responsive to receiving the request, for switching the call to a second terminal associated with the user, wherein the second terminal uses the circuit switched network and wherein the call is switched to the second terminal without terminating the call;

wherein the first terminal has a first path to a switching node and a second path from the switching node to a destination node and wherein the switching means comprises:

creating means for creating a third path to the second terminal on the circuit switched network; and

joining means for joining the third path to the second path, wherein the second terminal is joined into the call.

21. The communications system of claim 20 further comprising:

joining means for joining the third path to the first path, wherein the second terminal is joined into a conference call.

22. The communications system of claim 21, wherein the first path, the second path, and the third path are joined at a conference bridge in the switching node.

23. A communications system for routing a call, the communications system comprising:

receiving means for receiving a request from a user, at a first terminal, during a call to switch the call from a circuit switched network to a packet based network; and

switching means, responsive to receiving the request, for switching the call to a second terminal associated with the user, wherein the second terminal uses the packet based network and wherein the call is switched to the second terminal without terminating the call.

24. The communications system of claim 23, wherein the initiating means comprises:

establishing means for establishing a path to the second terminal on the packet based network; and

using means, responsive to an answer at the second terminal, for using the path to continue the call without interruption.

25. A computer program product in a computer readable medium for routing a call, the computer program product comprising:

first instructions for receiving a request from a user, at a first terminal, during a call to switch the call from a circuit switched network to a packet based network; and

second instructions, responsive to receiving the request, for switching the call to a second terminal associated with the user, wherein the second terminal uses the packet based network and wherein the call is switched to the second terminal without terminating the call.

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US005615253A

United States Patent [19]

Kocan et al.

[11] **Patent Number:** **5,615,253**

[45] **Date of Patent:** **Mar. 25, 1997**

[54] **METHOD FOR PROCESSING FORWARDED TELEPHONE CALLS**

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[73] Assignee: **AT&T**, Middletown, N.J.

[21] Appl. No.: **330,418**

[22] Filed: **Oct. 28, 1994**

[51] **Int. Cl.**[6] **H04M 1/66**

[52] **U.S. Cl.** **379/196**; 379/189; 379/210; 379/211; 379/243

[58] **Field of Search** 379/188, 189, 379/190, 191, 192, 193, 194, 195, 196, 197, 198, 210, 211, 212, 207, 243, 249

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Assistant Examiner—Daniel S. Hunter
Attorney, Agent, or Firm—Mark K. Young

[57] **ABSTRACT**

Increased network security is provided in accordance with the invention by using information obtained from the signaling network to determine whether a call has been forwarded, and then using this information to make a determination as to appropriate further call processing, to minimize a communications company's exposure to call forwarding fraud. A determination of whether a call is a forwarded call can be made on the basis of a call forwarding indicator provided by another portion of the communications network. Where such an indicator is unavailable, a determination of whether a call is a forwarded call can be made by comparing the telephone number dialed to originate the call with the telephone number (the "connect number") associated with the telephone station to which the call has been forwarded. When the dialed number and the connect number differ, the call is determined to be a forwarded call.

8 Claims, 3 Drawing Sheets

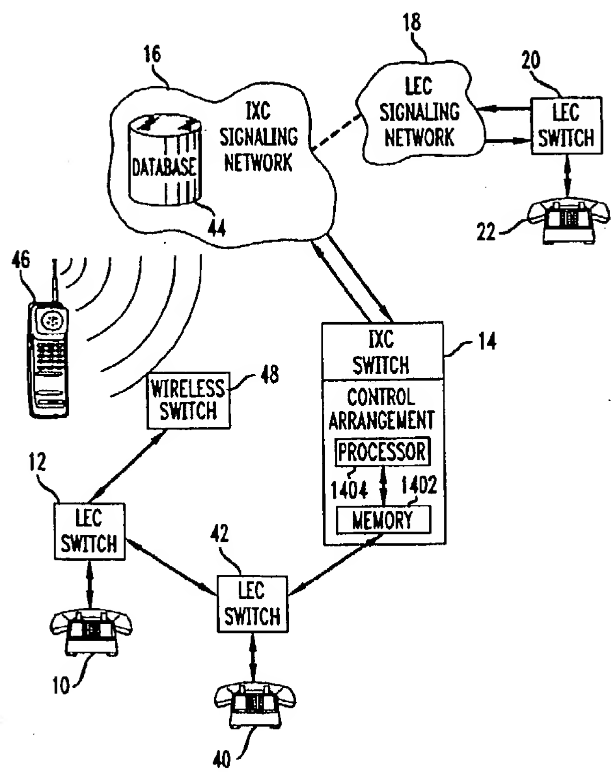


FIG. 1

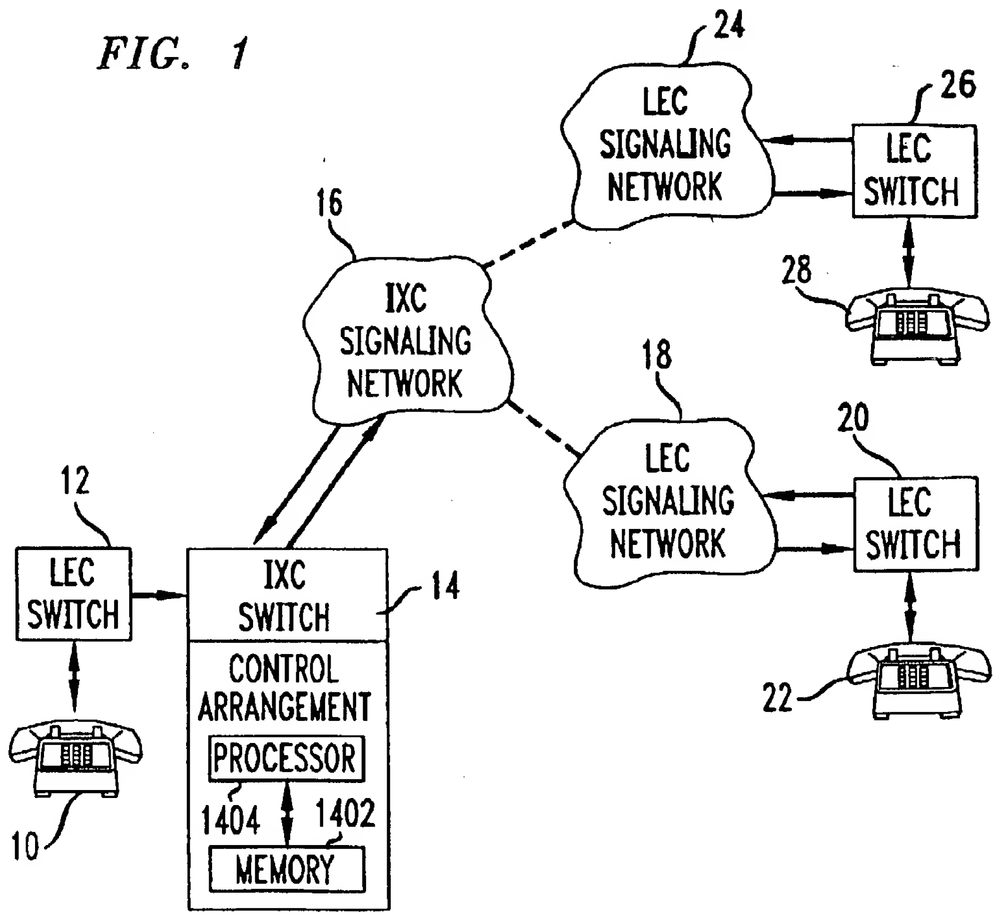

24
LEC SIGNALING NETWORK
26
LEC SWITCH
28
16
IXC SIGNALING NETWORK
18
LEC SIGNALING NETWORK
20
LEC SWITCH
22
12
LEC SWITCH
10
IXC SWITCH
14
CONTROL ARRANGEMENT
PROCESSOR
1404
1402
MEMORY

FIG. 2

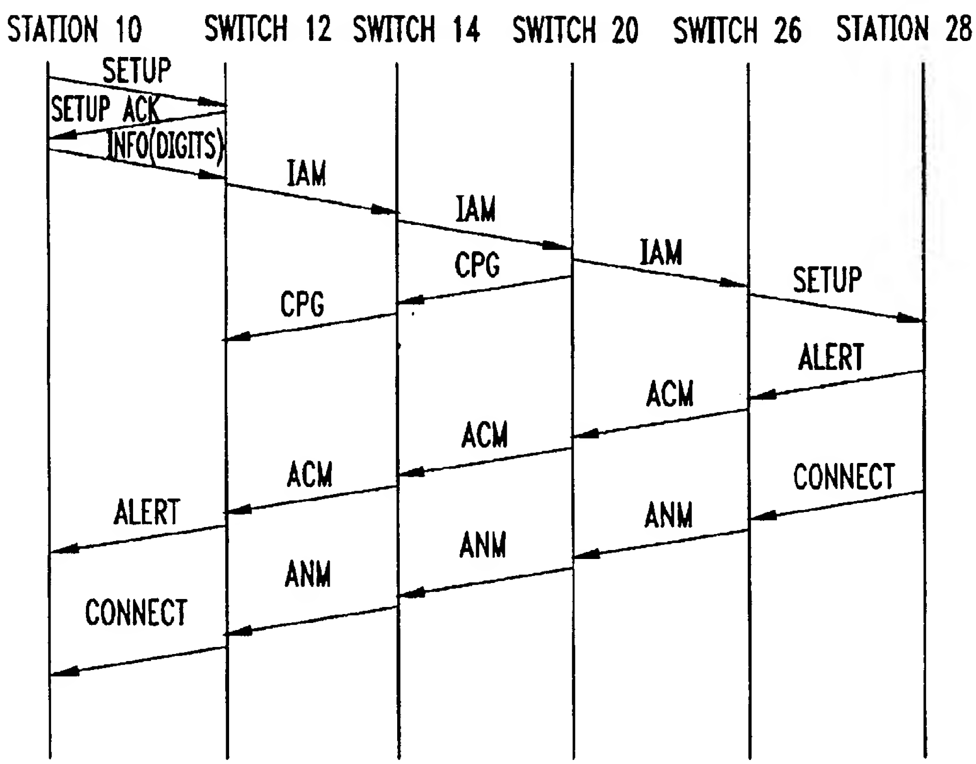
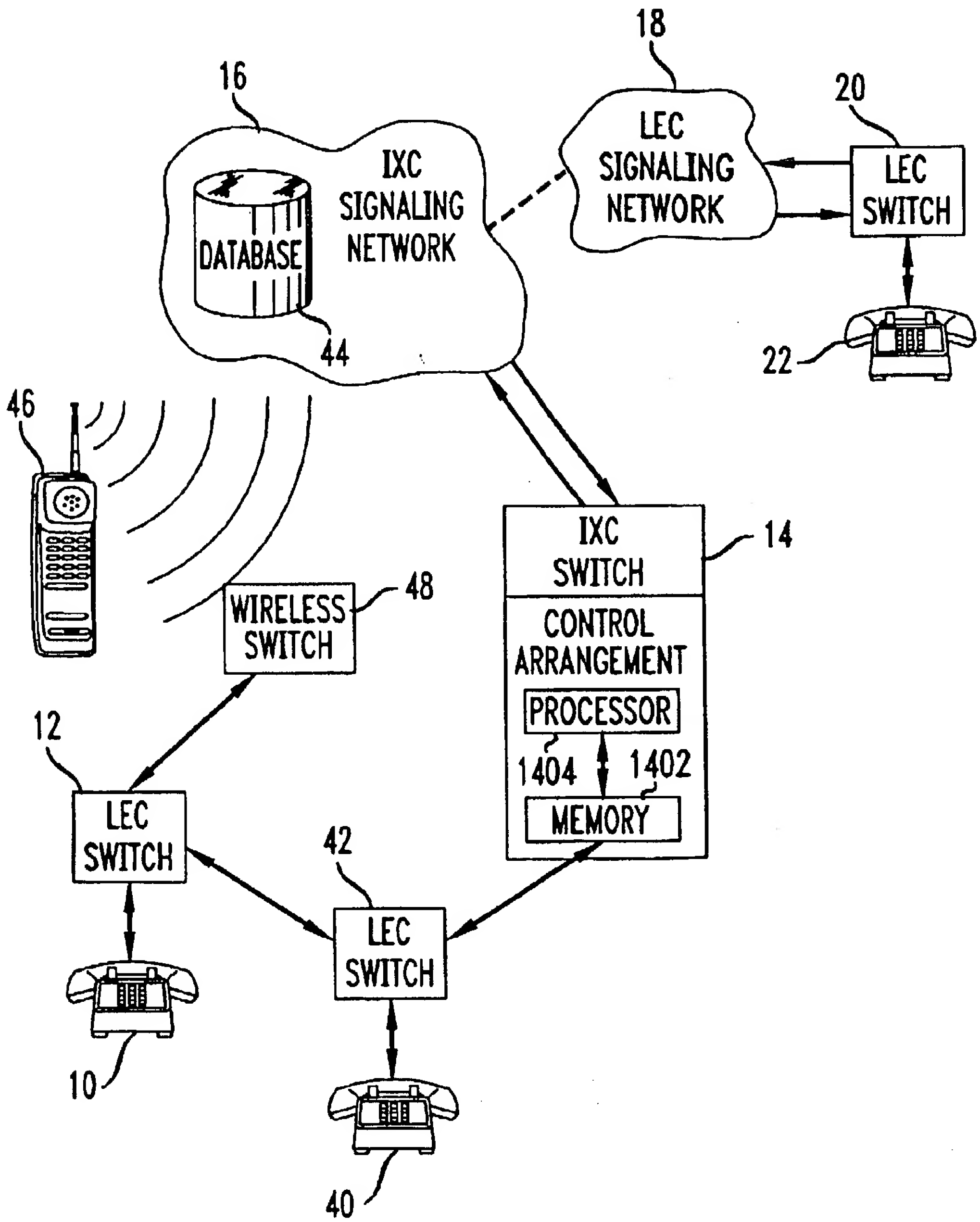

STATION 10
SWITCH 12
SWITCH 14
SWITCH 20
SWITCH 26
STATION 28
SETUP
SETUP ACK
INFO(DIGITS)
IAM
IAM
IAM
SETUP
CPG
CPG
ALERT
ACM
ACM
ACM
CONNECT
ALERT
ANM
ANM
ANM
CONNECT

FIG. 3



16
IXC
SIGNALING
NETWORK
DATABASE
44
18
LEC
SIGNALING
NETWORK
20
LEC
SWITCH
22
46
WIRELESS
SWITCH
48
IXC
SWITCH
14
CONTROL
ARRANGEMENT
PROCESSOR
1404
1402
MEMORY
12
LEC
SWITCH
42
LEC
SWITCH
10
40

METHOD FOR PROCESSING FORWARDED TELEPHONE CALLS

TECHNICAL FIELD

This invention relates to methods for processing telephone calls which have been forwarded from one telephone station to another and, more particularly, to processing forwarded telephone calls to minimize telephone fraud.

BACKGROUND OF THE INVENTION

Call forwarding is a telephone feature which allows a customer to direct a communications network to re-route telephone calls from one location to another location. Specifically, calls placed to a dialed number are re-routed to a telephone station identified by a different telephone number specified by the customer when setting up the call forwarding feature. Call forwarding, however, is susceptible to various telecommunications fraud schemes. In particular, persons attempting to defraud the telephone company (referred to hereafter as "hackers") subscribe, either legitimately or fraudulently, to telephone service with call forwarding as a service feature. The hackers then arrange to place calls to telephone numbers, using the call forwarding feature, which would otherwise be blocked by the network.

Current methods for protecting and preventing unauthorized use of the communications network have not adequately addressed the problem. For example, methods which detect fraud based on data obtained at the end of one or more billing cycles do not provide sufficiently timely information. By the time the information becomes available to indicate fraud, large amounts of fraudulent usage could already have occurred. Operator assisted calls involve further difficulties in detecting and blocking fraud because anti-fraud protections may be bypassed.

SUMMARY OF THE INVENTION

Increased network security is provided in accordance with the invention by using information obtained from the signaling network to determine whether a call has been forwarded, and then using this information to make a determination as to appropriate further call processing, to minimize a communications company's exposure to call forwarding fraud. A determination of whether a call is a forwarded call can be made on the basis of a call forwarding indicator provided by another portion of the communications network. Where such an indicator is unavailable, a determination of whether a call is a forwarded call can be made by comparing the telephone number dialed to originate the call with the telephone number (the "connect number") associated with the telephone station to which the call has been forwarded. When the dialed number and the connect number differ, the call is determined to be a forwarded call.

In an exemplary embodiment of the invention, the connect number is compared with the dialed number and, where the two numbers differ, either the call is terminated or preselected preventative action is initiated within the network. This embodiment is useful, for example, in the context of processing telephone calls dialed from a prison. In another embodiment of the invention, the connect number is subjected to the same fraud screening process that is applied to the dialed number. For example, if the dialed number is subject to geographical dialing restrictions, such as being limited to calls within the United States, the connect number also must be a number within the United States or the call is terminated.

BRIEF DESCRIPTION OF THE DRAWINGS

In the drawings:

FIG. **1** is a simplified block diagram of a portion of a telecommunications network, including signaling network components, suitable for processing outbound forwarded calls in accordance with the present invention;

FIG. **2** is an illustrative message sequence diagram for setting up an outbound forwarded call; and

FIG. **3** is a simplified block diagram of a portion of a telecommunications network, including signaling network components, suitable for processing inbound forwarded calls in accordance with the present invention.

DETAILED DESCRIPTION

Before describing the novel aspects of the invention, it will be useful to describe the path through an illustrative communications network of a typical forwarded call. Interexchange calls (e.g., inter-lata and international calls) can be forwarded in at least two different ways. First, the call can be routed through the interexchange carrier (IXC) and then forwarded by a local exchange carrier (LEC) switch. Such a call is referred to herein as an "outbound" forwarded call. Alternatively, the call can be forwarded by a LEC switch and then routed through the IXC. Such a call is referred to herein as an "inbound" forwarded call. A different signaling operation takes place depending on whether the call is an outbound or inbound forwarded call. Each type of call will be discussed below.

Referring now to the drawings, FIG. **1** can be used to illustrate the signaling which occurs to set up an outbound forwarded call. FIG. **1** shows a portion—particularly the signaling portion—of an exemplary communications network at telephone station **22** subscribes to a call forwarding service and has arranged. The network includes telephone stations **10**, **22**, and **28**, LEC switches **12**, **20**, and **26**, IXC switch **14**, IXC signaling network **16**, and LEC signaling networks **18** and **24**. It is assumed that a calling party at a telephone station **10** desires to place a call to a party at a telephone station **22**, and that the party to have all calls forwarded to telephone station **28**. LEC switches **12**, **20**, and **26** may be, for example, a 5ESS® switch, which is commercially available from AT&T Corp. IXC switch **14** may be, for example, a 5ESS® switch or a 4ESS™ switch (also commercially available from AT&T Corp.).

LEC switches **12** and **26** communicate with the other switches in the call path by exchanging call handling messages via a data network called a Common Channel Signaling (CCS) network. The CCS network, shown in part in FIG. **1** as IXC signaling network **16**, is a packet switching network having a plurality of interconnected nodes called Signal Transfer Points (STPs) that are used to exchange call handling messages between switches according to a specific protocol, such as CCS7. However, for the sake of simplicity, the constituent elements of the signaling network are not expressly shown. The features and functionality of an STP are described in the book *Engineering and Operations in the Bell System, Second Edition,* AT&T Bell Laboratories, 1992, pp. 292–294.

The invention will be described herein in the context of messages using the ISDN User Part (ISUP) protocol. ISUP is an interoffice protocol for circuit-related functions that interworks with Q.931 signaling. ISUP supports calls between ISDN subscriber for basic bearer services and supplementary services (such as Call Forward Busy (CFB),

Call Forward No Reply (CFNR), and Call Forward Unconditional (CFU)) for voice and non-voice applications in an ISDN. However, ISUP also supports calls between non-ISDN subscribers. The ISUP message is generated and interpreted by the switches of the CCS network and is carried as the user data in the MTP or SCCP message. LEC switches **12** and **26** communicate with telephone stations **10** and **28**, respectively, using a conventional signaling arrangement for the control of circuit-switched calls, illustratively Q.931 signaling. ISUP, with the use of the CCS network, extends Q.931 (which is a point-to-point network access protocol) over a store-and-forward message switching network. While the invention is discussed in the context of CCS7, ISUP, and Q.931 signaling, one skilled in the art will readily appreciate that the principles of the invention are not limited by the type of network signaling used, but rather are applicable to any switching system in which the dialed number, connect number, and call forward indicators can be captured when a call, or a leg of a call, is being set up.

FIG. 2 shows exemplary signaling used to set up an outbound forwarded call. When a call is initiated from telephone station **10** to telephone station **22**, switch **12** collects dialed digits from telephone station **10** using well known stimulus signaling methods. Telephone station **10** sends a SETUP message (using Q.931 signaling) to switch **12**. Switch **12** then returns a SETUP ACK message to station **10**. The caller at station **10** then enters the destination directory number (dialed number) and station **10** transmits a sequence of info messages each including one or more digits of the dialed number. (Alternatively, all of the dialed number digits may be included in the setup message.) The SETUP message also includes calling party identification information, such as the originating telephone number, or automatic number identifier (ANI). Switch **12** uses the received dialed number to generate and transmit an ISUP Initial Address Message ("IAM") to switch **14**. Switch **14**, in turn, transmits an IAM message via signaling networks **16** and **18** to LEC switch **20**. LEC switch **20** recognizes that telephone station **22** has activated the call forwarding feature. Rather than setting up the call with telephone station **22**, LEC switch **20** transmits an IAM message to LEC switch **26** via signaling networks **18**, **16**, and **24** to effect the call forwarding service. At the same time, LEC switch **20** transmits an ISUP Call Progress Group ("CPG") message to IXC switch **14** to notify the switch that the call has been forwarded. LEC switch **26** transmits a SETUP message to telephone station **28** to set up the call.

At this point, call processing continues in a conventional manner by transfer of signaling messages between telephone station **28** and telephone station **10**. Station **28** sends an ALERT message to switch **26**. Switch **26** transmits an Address Complete Message (ACM) via switches **20** and **14** to switch **12**. Switch **12** then sends an ALERT message to station **10**. Telephone station **28** then sends a Connect message to switch **26**. Switch **26** sends an Answer Message (ANM) call supervision message via switches **20** and **14** to switch **12**. Switch **12** then sends a Connect message to telephone station **10**.

FIG. 3 shows a network for switching an inbound forwarded call. In addition to various elements common to those shown in FIG. 1, the network of FIG. 3 includes a telephone station **40**, a LEC switch **42**, and a database **44** that is disposed within signaling network **16**. (FIG. 3 also includes a mobile telephone station **46** and a wireless switch **48**, which will be described later.) In this example, call forwarding is effected before the call reaches the IXC network. In particular, a caller at telephone station **10** places a call to telephone station **40**. Telephone station **40** is arranged to forward calls to telephone station **22**. Telephone station **22** is located in a different area code or country than telephone stations **10** and **40**.

One skilled in the art will appreciate that the signaling messaging for establishing an inbound forwarded call is similar in nature to the messaging for outbound forwarded calls shown in FIG. 2. When a call is initiated from telephone station **10** to telephone station **40**, telephone station **10** sends a Setup message to switch **12**, causing the switch to collect dialed digits from telephone station **10**. Switch **12** then transmits an IAM message to switch **42**. LEC switch **42** recognizes that telephone station **40** has activated the call forwarding feature to forward calls to telephone station **22**. Rather than setting up the call with telephone station **40**, LEC switch **42** transmits an IAM message to IXC switch **14**, which, in turn, transmits an IAM message via signaling networks **16** and **18** to LEC switch **20** to effect the call forwarding service. LEC switch **20** then transmits a Setup message to telephone station **22** to set up the call. At this point, call processing continues in a conventional manner by transfer of signaling messages between telephone station **22** and telephone station **10**. In setting up the inbound forwarded call, LEC switch **42** typically will indicate to IXC switch **14** that the call is a forwarded call. This notification may be provided by using a call forwarding indicator, for example, by using ISUP parameters 3.20 or 3.25.

We have recognized that by monitoring the call set up process, information transferred via the signaling network as part of the signaling used to set up the call can be captured and used to detect and prevent telephone fraud effected through the use of the call forwarding service. In particular, information captured is used to determine whether a telephone call has been completed to the dialed number. The dialed number and the connect number are captured by various components of the signaling network at various times during call set up. The dialed number and the connect number are then compared or otherwise analyzed to determine the manner in which the call is to be processed. As discussed below, the manner in which this information is collected may depend upon whether the call is an outbound or inbound forwarded call.

In the case of an outbound forwarded call, as in FIG. 1, the call progress (CPG) message transmitted from LEC switch **20** to IXC switch **14** includes an indication of the telephone number (e.g., the ANI) of the telephone station to which the call was forwarded, together with a call forward indicator. IXC switch **14** captures this information in a memory location **1402** for subsequent processing. Switch **14** also stores the originally dialed number, which it received as part of the IAM message from LEC switch **12**. A processor **1404**, illustratively disposed within IXC switch **14**, communicates with memory location **1402** to process the dialed number, connect number, and call forward indicator in accordance with the invention. In the case of an inbound forwarded call, as in FIG. 3, IXC switch **14** captures the dialed number, connect number, and call forward indicator from signaling messages received from LEC switch **42** and stores the numbers in memory location **1402**. It is to be understood that processor **1404** and memory location **1402**, or the functionality of these elements, could be disposed within a LEC switch, an operator position, or a signaling network component such as an STP.

Although it would be preferable to capture as much information as possible from the signaling messages to determine whether a forwarded call represents an attempt to defraud the communications carrier, it is to be understood

that only part of the information may be available for capture. For example, only a call forward indicator may be available, or only the dialed and connect numbers may be available. One skilled in the art will readily appreciate, in view of this disclosure, that collecting part of the information often will suffice to permit a determination of whether to block, terminate, or otherwise track and process a call suspected of being fraudulent.

Having described how to collect the dialed number, connect number, and call forward indicator for a forwarded call, various ways in which the information can be used to minimized fraud will now be described. There are many different ways in which the dialed number and connect number can be used in this context, a few examples of which are discussed in turn below.

Call termination

First, the information that a call has been forwarded can be used to determine whether to complete the call. For example, calls for which the dialed number and connect number are different, or for which a call forward indicator is present, simply are terminated. As used herein, "terminating" a call refers to preventing a normal voice path (or data path for facsimile calls and other data transmissions), and includes blocking of the call before the path is established or tearing down the call if the path has already been established. Processor **1404** (FIG. **1**) retrieves the dialed number and the connect number (i.e., the "forwarding telephone number") from memory **1402**, compares the two numbers, and signals switch **14** to terminate the call when the two numbers are different (or upon another suitable analysis). Similarly, the presence of a call forward indicator can be used by processor **1404** to initiate call termination. This may be achieved, for example, by causing switch **14** to transmit a Release call supervision message to the other switches involved in the call, preferably before LEC switch **12** sends the Connect message to station **10** (see FIG. **2**). This type of processing may be appropriate where call forwarding simply is not allowed for the call. One example where such processing would be appropriate is the limited telephone service available to prisoners in the criminal justice system—prisoners are allowed to call only selected telephone numbers. By restricting the use of call forwarding entirely, prisoners are prevented from placing unauthorized calls via the call forwarding mechanism which would have been blocked had the call been directly dialed to that destination. Because calls placed from cellular or other radio-based telephones are subject to high fraud, including fraud committed via call forwarding, it may also be appropriate to terminate all cellular calls connected to a number other than the dialed number.

Information indicating that a call has been forwarded also can be used to subject the forwarded call to the same terminating call restrictions that would have been applied to the dialed number in determining whether to complete the call. In other words, if the connect number does not satisfy the terminating call restrictions applicable to the dialed number, the call is blocked or some further fraud prevention activity is initiated. For example, if a caller at telephone station **10** (FIG. **1**) is restricted from directly dialing telephone numbers outside the United States, calls originating at telephone station **10** will not be forwarded to a telephone number corresponding to a destination outside the United States. In this manner, the caller is prevented from using the call forwarding service to "dial around" the terminating call restrictions on the telephone from which a call is placed.

In a further embodiment of the invention, a call for which the dialed number and connect number are routed to an attendant. Processor **1404** retrieves the dialed number and the connect number from memory **1402**, compares the two numbers, and routes the call to an attended operator position or other customer service attendant. The attendant may then question the caller to obtain further information demonstrating the caller's right to complete the call. The attendant then determines whether to complete or terminate the call.

In still another embodiment of the invention, forwarded calls are flagged for further investigation or processing. The further processing can take many forms. For example, once the call is identified as a forwarded call, either by call forwarding indicator or a difference between the dialed and connect numbers, processor **1404** can analyze predetermined call attributes and selectively terminate those calls having attributes indicative of fraudulent calls. If, for example, a given call is identified as a forwarded call, and processor **1404** determines that the connect number has a country code outside the United States, processor **1404** may automatically terminate the call.

Forwarded calls flagged for further investigation as described above can be processed according to the call forwarding history of the dialed number. That is, forwarded calls are checked against call detail records stored in a database, such as database **44** of FIG. **3**, to determine how often calls to that dialed number have been forwarded within some specified period of time. The database, which may be a network control point (NCP) commercially available from AT&T Corp., would store records having at least the dialed number and an indication of whether the call to the dialed number was forwarded to another number. The records preferably also would include the date and time of the call, the ANI of the originating telephone station, and the connect number. In operation, call processing would proceed as described above until processor **1404** determines that the call is a forwarded call. Upon detecting call forwarding, processor **1404** queries the database **44** with a message which includes the dialed number, a call forward indicator, and preferably the ANI of the originating telephone station, the connect number, and the date and time of the call. The database includes a processor under the control of suitable programming which, in response to the call forwarding indicator, compares the dialed number with the dialed number of the call detail records stored in the database. The processor of database **44** counts the number of occurrences (matches) in which the dialed number in the message received from switch **14** matches a record in the database having a dialed number and a call forwarding indicator. If the number of occurrences exceeds a predetermined threshold (as specified in fraud prevention software installed in the database processor), database **44** returns a message to switch **14** instructing the switch to terminate the call or initiate other fraud prevention activities. The information provided to database **44** in the original message from switch **14** is added as a record to the database **44** as a call detail record. The database can be designed to automatically discard old call detail records on a rolling basis as new call detail records are added. Database **44** may be dedicated to monitoring call forwarding fraud, but preferably is part of another system or has other functions and uses so as to make the system more efficient.

While the invention has been discussed in the context of wired telephone service, the principles of the invention are equally applicable to wireless telephone service, such as calls originating from a cellular telephone. With reference to FIG. **3**, assume for the purpose of discussion that a caller at wireless telephone **46** originates a telephone call to telephone station **40**, and that telephone station **40** has arranged to forward the call to telephone station **22** outside the United

States. The call is switched through wireless switch **48**, to LEC switch **12**, and then to LEC switch **42**. LEC switch **42** forwards the call to telephone station **22**. In accordance with the invention, switch **14** detects that the call has not been completed to the dialed number (i.e., to telephone station **40**). In response to this determination, switch **14**, under the control of processor **1404**, takes the appropriate action to minimize the likelihood of call forwarding fraud on the communications service provider. Upon determining that the call is a forwarded call, switch **14** may determine that the call originated from a wireless telephone, for example, by examining the automatic number identifier, or ANI, of the originating telephone station. As discussed above, the indication that the call is a forwarded call, together with the indication that the call originated from a wireless telephone, may be the basis for initiating immediate termination of the call. Switch **14** also may use these indications together with the country code of the connect number as the basis for allowing or terminating the call.

The call forwarding fraud prevention techniques of the invention can be useful where a caller is directly connected to an IXC, such as through an operator position. For example, the principles described herein are applicable to calls billed to a calling card or credit card, and to calls placed (completed) by an attendant on the caller's behalf. In this regard, the principles of the invention can be utilized to provide an attendant with an indication that a given call has been forwarded to the attendant. Such an indication will enable the attendant to recognize the call as a forwarded call and refuse to complete the call.

One skilled in the art will appreciate that various modifications can be made to the network and the call forwarding fraud detection process without departing from the scope of the invention. For example, while the invention has been described in the context of voice and data call, the principles of the invention are equally applicable to multimedia calls, such as video telephone calls. Also, the invention can be used to detect (and terminate) calls that are forwarded multiple times before being completed to a final destination telephone station. In such a case, the dialed number and the connect number will differ, or a call forwarding indicator will be provided, in the same manner as discussed above for a call that is forwarded once.

We claim:

1. A call processing method comprising the steps of:

capturing a telephone number dialed to originate a call to a predetermined destination;

capturing a second telephone number associated with the telephone station to which the call will ultimately be connected;

comparing the dialed telephone number with the second telephone number;

responsive to a determination that the first and second telephone numbers are the same, further extending the call toward the destination; and

responsive to a determination that the dialed telephone number and the second telephone number differ, terminating the call.

2. A call processing method comprising the steps of:

determining whether a call to a communication station is a forwarded call;

responsive to a determination that the call is a forwarded call, preventing completion of the call to the communication station; and

wherein the step of determining whether the call is a forwarded call comprises the steps of:

capturing a telephone number dialed to originate the call to a predetermined destination;

capturing a second telephone number associated with the telephone station to which the call will ultimately be connected; and

comparing the dialed number with the second telephone number to determine that the call is a forwarded call when the dialed and second telephone numbers differ.

3. A call processing method comprising the steps of:

determining whether a call to a communication station is a forwarded call;

responsive to a determination that the call is a forwarded call, determining whether a telephone number dialed to originate the call is subject to a call restriction;

testing a second telephone number, associated with the telephone station to which the call will ultimately be connected, against the call restriction; and

responsive to a determination that the second telephone number does not meet the call restriction, initiating fraud prevention activity in connection with processing the call.

4. The method of claim **3** wherein the determining step comprises comparing dialed telephone number with the second telephone number to determine whether the call is a forwarded call.

5. The method of claim **3** wherein the fraud prevention activity in connection with processing the call comprises terminating the call.

6. A call processing method comprising the steps of:

determining whether a call to a communication station is a forwarded call;

responsive to a determination that the call is a forwarded call, accessing a database to obtain information indicative of whether the call represents unauthorized use of the communications network.

7. A method of processing a telephone call, comprising the steps of:

determining whether the call is to be completed to a telephone number dialed to originate the call;

responsive to a determination that the call will not be completed to the dialed number, initiating fraud prevention activity in connection with processing the call; and

wherein the step of initiating fraud prevention activity comprises the steps of:

determining whether the dialed telephone number is subject to a call restriction;

testing a second telephone number associated with the telephone station to which the call will ultimately be connected against the call restriction; and

terminating the call if the second telephone number does not meet the call restriction.

8. A method of processing a telephone call, comprising the steps of:

determining whether the call is to be completed to a telephone number dialed to originate the call;

responsive to a determination that the call will not be completed to the dialed number, initiating fraud prevention activity in connection with processing the call; and

wherein the step of initiating fraud prevention activity comprises accessing a database to obtain information indicative of whether the call represents unauthorized use of the communications network.

* * * * *

X

US005583564A

United States Patent [19]

Rao et al.

[11] **Patent Number:** 5,583,564

[45] **Date of Patent:** Dec. 10, 1996

[54] **INTELLIGENT CALL FORWARDING WITH VIDEOPHONE DISPLAY OF FORWARDING DESTINATION**

[75] Inventors: **Usha Rao**, Aberdeen; **Robert M. Rubin**, Morristown, both of N.J.

[73] Assignee: **Lucent Technologies Inc.**, Murray Hill, N.J.

[21] Appl. No.: **427,538**

[22] Filed: **Apr. 24, 1995**

[51] **Int. Cl.**[6] **H04N 7/14**; H04M 3/54

[52] **U.S. Cl.** **348/14**; 379/207; 379/210; 379/211

[58] **Field of Search** 379/96, 201, 207, 379/210, 211, 212; 348/15, 14

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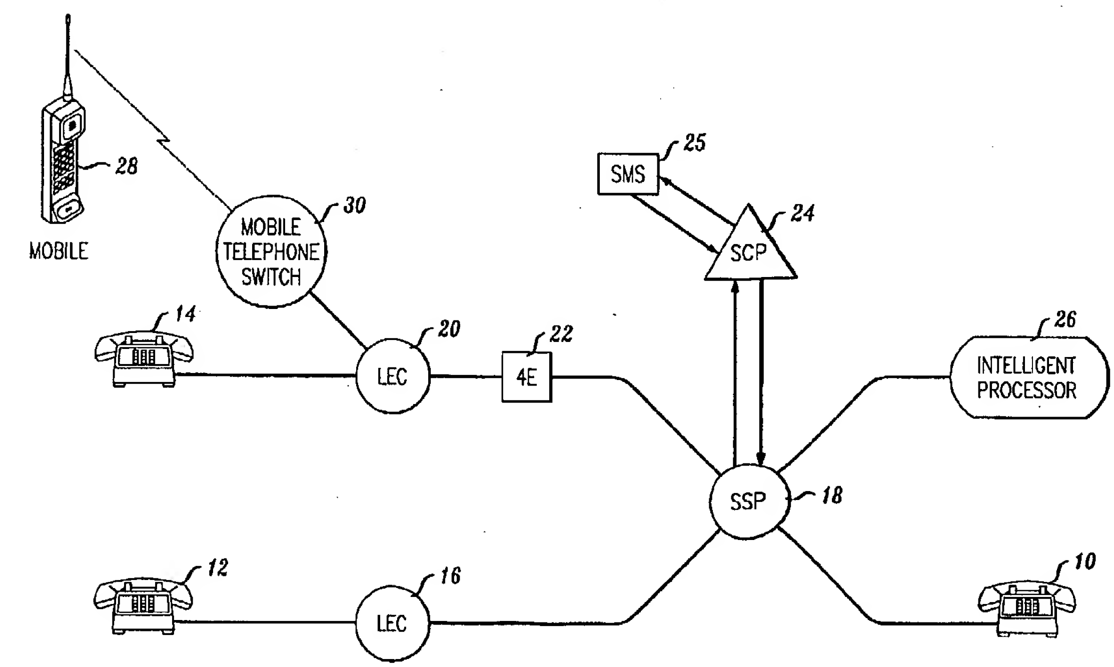
Primary Examiner—Krista M. Zele
Assistant Examiner—Daniel S. Hunter

[57] **ABSTRACT**

Intelligent call forwarding is implemented through an intelligent network. The intelligent network includes a Service Switching Point (SSP), a Service Control Point (SCP), and a Service Management System (SMS). A call to a network subscriber is routed to the SSP which is used to detect Call Forwarding "triggers"—conditions which require the SSP to initiate selective Call Forwarding service. When a Call Forwarding trigger is detected the SSP passes the calling party's number to the SCP and requests call handling instructions from the SCP. The SCP provides the call handling instructions, as provisioned by the subscriber, to the SSP, and the SSP then forwards the call in accordance with the instructions. The SMS formulates and sends commands to the SCP for such purposes as changing the stored subscriber instructions or updating service features. By provisioning the SCP with appropriate instructions the forwarding destination for a call may be made to depend upon the time of day, day of the week, caller's identity, and/or status of the called telephone.

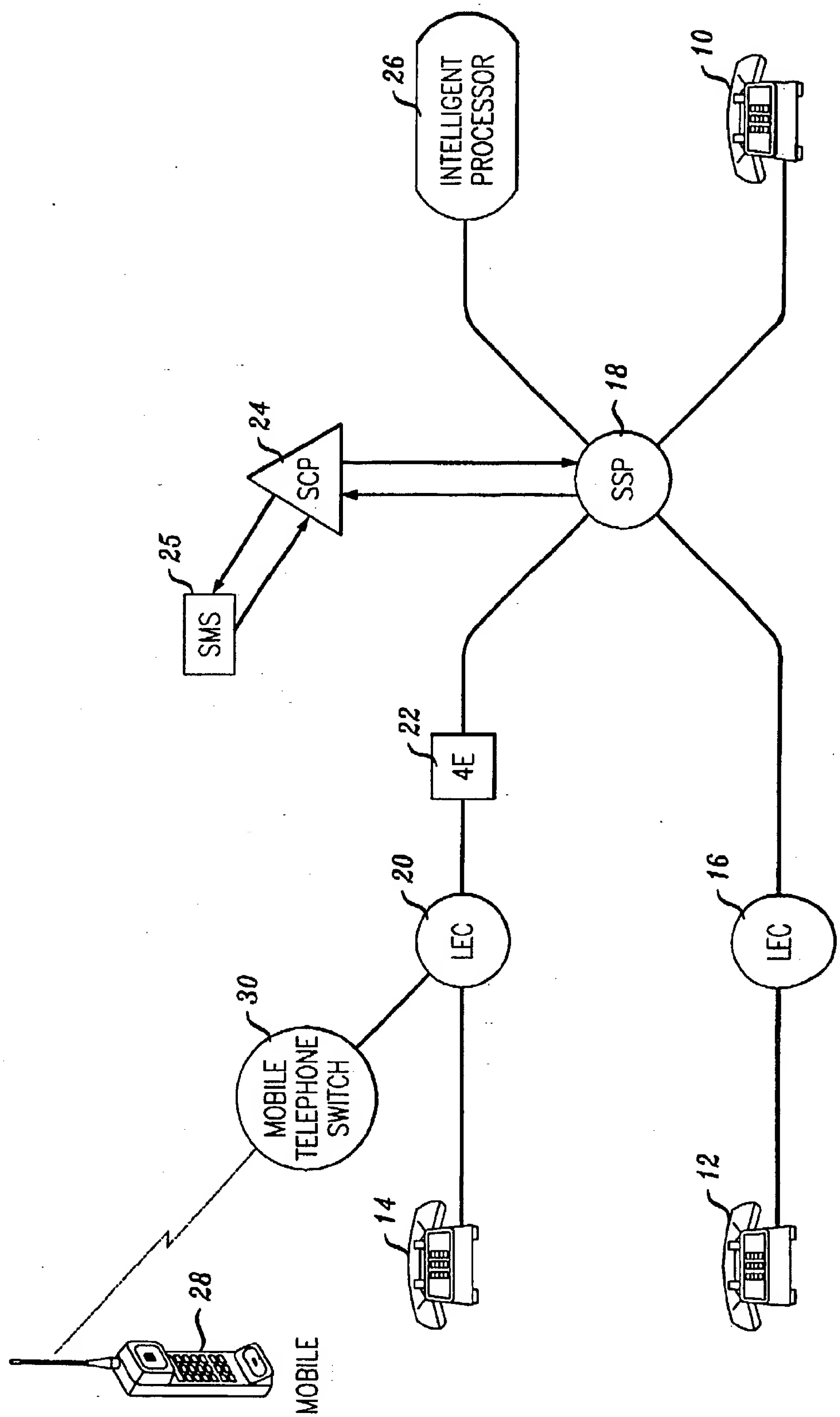
pre-programmed

12 Claims, 1 Drawing Sheet



instructions program

02/02/2004, EAST Version: 1.4.1


INTELLIGENT PROCESSOR
26
10
18
SSP
24
SCP
25
SMS
22
4E
20
LEC
16
LEC
30
MOBILE TELEPHONE SWITCH
14
12
28
MOBILE

INTELLIGENT CALL FORWARDING WITH VIDEOPHONE DISPLAY OF FORWARDING DESTINATION

TECHNICAL FIELD

This invention relates to communication networks, and more particularly to providing call forwarding service to communication network subscribers.

BACKGROUND OF THE INVENTION

Establishing communication with a called party becomes difficult when that party is mobile. In general, a person wishing to contact a party by telephone initiates a call to a telephone that is in close proximity to that party, such as a home telephone or office telephone. However, the called party may not always be in close proximity to any one particular telephone at all times. Moreover, highly mobile individuals may be located near several different telephones during the course of a day and, thus become difficult to contact without a priori information of their whereabouts. This presents a problem to persons who need to contact a mobile party immediately, regardless of that party's location.

An attempt to solve the problem of establishing communication with a mobile party resulted in the technique known as "Call Forwarding". Call Forwarding is a well-known technique whereby an individual who will be away from her telephone can redirect calls to another telephone. For example, by using Call Forwarding an office worker planning to go on vacation could redirect her calls to an office mate's telephone. Such systems, although desirable, are dramatically limited by their inability to selectively forward calls. Once a subscriber has specified a forwarding number to which the subscriber's calls should be forwarded, all of the subscriber's calls are forwarded to that number, regardless of the time of day, day of the week, identity of the calling party, or status of the telephone to which calls are being forwarded.

SUMMARY OF THE INVENTION

The problems of prior Call Forwarding systems are overcome by employing an intelligent network configured to selectively forward calls.

The intelligent network includes a Service Switching Point (SSP), a Service Control Point (SOP), and a Service Management System (SMS). A call to a network subscriber is routed to the SSP which is used to detect Call Forwarding "triggers"—conditions which require the SSP to initiate selective Call Forwarding service. When a Call Forwarding trigger is detected the SSP passes the subscriber's (called party's) number to the SOP and requests call handling instructions from the SOP. The SOP provides the call handling instructions, as provisioned by the subscriber, and the SSP forwards the call in accordance with the instructions. The SMS formulates and sends commands to the SOP for such purposes as changing the stored subscriber instructions or updating service features.

Through the use of the SSP, SCP, SMS, and their associated triggering mechanism a forwarding destination for a call may be made to depend upon the time of day, day of the week, caller's identity, and/or status of the called telephone.

BRIEF DESCRIPTION OF THE DRAWING

FIG. 1 is a simplified block diagram of a network, including intelligent network components, suitable for use with the present invention.

DETAILED DESCRIPTION

Before describing an exemplary embodiment of the invention, it will be useful to describe in a general manner the key intelligent network elements which can be used to implement the invention. Intelligent network components suitable for implementing the invention, in view of this disclosure, are well-known in the art and are commercially available from the AT&T Corporation ("AT&T") under the A-I-Net™ advanced intelligent network family of products.

The intelligent network architecture superimposes on an existing telecommunications system a modular configuration of network elements which provide enhanced telecommunications services. Switching functions are performed by the base network in a conventional manner. The intelligent network includes a service switching point (SSP), a service control point (SCP), and a service management system (SMS). The intelligent network may also include an intelligent processor. One skilled in the art will appreciate that the intelligent network elements could be owned or controlled by a local exchange carrier (LEC), an interexchange carrier (IXC), a competitive access provider, or some combination of the three.

The SSP is a switch that operates to recognize service requests, requests call handling instructions from an SCP, and executes those instructions to complete a telephone call. The SSP provides intelligent network "triggering"—detecting a condition which requires the SSP to initiate the intelligent network service by sending a query to the SCP. As described below, the intelligent call forwarding service of the invention has its own "trigger profile," or set of data, that assigns the service a unique point of entry into intelligent network functions. The SSP also formulates and transmits requests to the SCP and processes replies and requests from the SCP. In addition, the SSP creates and plays intelligent network announcements formulated by the service provider (e.g., the local exchange or inter-exchange carrier), and transmits event messages (such as busy or no reply signals) to the SCP. The SSP illustratively is an AT&T 5ESS® switch provisioned with AT&T's A-I-Net™ intelligent software to provide SSP functionality.

It should be noted that the SSP is capable of functions other than those mentioned above, such as processing billing records for a call. However, these "other functions" are beyond the scope of this invention, and therefore will not be described in detail.

The SCP is an intelligent network element which stores call control and call routing instructions to be executed by an SSP. It receives and processes event messages from the SSP, and formulates and sends responses to the SSP. In addition, the SCP processes accounting and statistical information, such as the number of the calling party, the dialed intelligent network number, the time of day and day of the week of the call, and various other call parameters. An example of an SCP capable of implementing the foregoing functions is AT&T's A-I-Net™ SCP. The SCP interfaces with and receives commands for controlling services and service features from the SMS.

The SMS is a management and provisioning system that serves as an intelligent network service administration platform. The SMS formulates and sends commands to the SCP

to control services and service features. An example of an SMS is the AT&T A-I-Net™ service management system.

The intelligent processor provides specialized functionality, such as speech recognition (identifying spoken words) and voice recognition (recognizing the voice of a particular speaker) capability. The intelligent processor also may perform the functions of a video signal generator or video signal database for applications such as providing images for use in a video telephone call forwarding service. The functionality of the intelligent processor may be implemented in a separate network element, or may be implemented through a multimedia SCP. As mentioned above, examples of services which may be provided by the intelligent processor are speach recognition and voice recognition. Other examples of services that may be offered through the intelligent processor include voice digit dialing and name dialing. Intelligent processor equipment suitable for use with the invention is well known in the art of intelligent network systems.

One area of intelligent call processing that merits special attention is the concept of "triggering." Triggering is the process by which a switch (e.g., an SSP) determines that a query message requesting call processing instructions will be sent to an SCP. A trigger is an occurrence of an event and/or the satisfaction of certain conditions which results in a message to the SCP. Triggers can be originating triggers, mid-call triggers, or terminating triggers. Examples of originating triggers are off-hook immediate, offhook delay triggers, and custom dialing plan triggers. An example of a mid-call trigger is the busy condition. An example of a terminating trigger is the ring-no answer condition. In the intelligent call forwarding method of the present invention distinct call forwarding routines may be initiated in response to each trigger.

Having described in a general manner the function of the principal intelligent network elements, an exemplary network architecture suitable for implementing the present invention will now be described. In the description references will be made to FIG. **1**.

Assume for purposes of illustration that the called party at a first telephone station **10** is busy on a call with a party at a second telephone station **12**, and that there is an incoming call for telephone station **10** from a third telephone station **14**. The call between telephone stations **10** and **12** extends from telephone station **12** to a LEC switch **16**, and is routed through an SSP **18** to the called party at telephone station **10**. The call from telephone station **14** is extended to a LEC switch **20** and routed via an IXC switch **22** to SSP **18**.

At this point, it should be noted that there are many well known means for coupling telephone calls between telephone stations. Examples of such coupling means, all of which are suitable for use in the present invention, are: standard telephone lines, twisted shielded pair lines, coaxial cables, fiber optic lines, and wireless links. It should also be noted that various types of "calls" and various types of "telephone stations" fall within the scope of the invention. For example, a "call" may take the form of a data transmission from a computer or a fax transmission from a facsimile machine. Moreover, "telephone stations" to which calls are forwarded may include dedicated answering machines, such as a voice mailbox.

In any event, SSP **18** functions as described above. When the call from telephone station **14** is routed to SSP **18**, the SSP attempts to route the call to telephone station **10**. SSP **18** detects that telephone station **10** is busy on another call, which produces a mid-call trigger. That is, the busy condition at telephone station **10** triggers SSP **18** to send a query, the called telephone number, and the calling telephone number (i.e., the ANI of telephone station **14**) to an SCP **24**. Based on this information, the SCP accesses a database and retrieves a set of call processing instructions that correspond to the call forwarding service prearranged, or "subscribed to", for telephone station **10**. The SCP may then direct the SSP to carry out one or more commands that are set forth in the instructions.

Communications between the SSP and SCP may be carried out over a Common Channel Signaling (CCS) network. CCS networks are well known in the art of telecommunications. They are generally used to communicate call control information among network elements, and they typically employ packet switching techniques to accomplish this task. The packet switches used in CCS networks are commonly referred to as Signal Transfer Points (STPs). In one possible implementation of a CCS, STPs are used in conjunction with CCS data links to pass call control information via a Signaling System 7 (SS7) protocol.

Regardless of whether or not a CCS is used to couple the SSP and SCP, the call forwarding operation proceeds as follows. SCP **24** notifies SSP **18** to play appropriate announcements to the caller at telephone station **14** and to expect additional inputs (e.g., Dual Tone Multi Frequency (DTMF) digits or voice commands) from the caller. SSP **18** plays an announcement to the caller requesting the caller's Personal Identification Number (PIN) and collects digits from the caller. SSP **18** then forwards the digits to SCP **24**, which determines the appropriate call processing on the basis of the caller-provided information. This may be, for example, to forward the caller to a voice messaging system and indicate to telephone station **10** that the caller is leaving a message. However, one of ordinary skill in the art will appreciate that the appropriate call processing may be determined on the basis of the ANI rather than on the basis of the caller-provided information.

SCP **24** is provisioned to provide call processing instructions, passed to it from an SMS **25**, to SSP **18**. SSP **18** then executes the instructions provided by the SCP. As described above, one possible set of instructions includes: collecting a PIN number from the caller, using the PIN to determine a forwarding destination for the caller, and forwarding the caller to the forwarding destination. Other possible sets of instructions allow for the forwarding destination to be dependent on the time of day and/or day of the week. To illustrate time/date dependent forwarding, mid-call triggering will again be considered.

The time/date dependent forwarding of a call may proceed as follows. A call from telephone station **14** is routed to SSP **18** which attempts to route the call to telephone station **10**. SSP **18** detects that telephone station **10** is busy on another call, resulting the production of a mid-call trigger on the busy condition. That is, the busy condition at telephone station **10** triggers SSP **18** to query an SCP **24** for call processing instructions. The SCP then instructs SSP **18** to collect a PIN from the caller, as in the previous example. The time of day and/or day of week may be collected from the caller in the same manner that the PIN was collected, or it may be retrieved from a clock. In either case, once the time/date information is collected, it may be used in conjunction with the PIN to determine a forwarding destination in accordance with the called party's (subscriber's) instructions.

In both the foregoing caller identification based forwarding and time/date based forwarding illustrations, the call is forwarded in response to a mid-call trigger. An example of

call forwarding in response to a terminating trigger is described below.

In the ring-no answer scenario, a call from telephone station **14** is routed to SSP **18**, which attempts to route the call to telephone station **10**. When there is no answer at telephone station **10**, a terminating trigger is generated and the SSP queries the SCP for call processing instructions. At this point, the call may be handled as it was in the busy condition case. That is, the call may be forwarded on the basis of the caller's identity and/or on the basis of the time/date; or, in the alternative, the call may be unconditionally forwarded to a predetermined destination.

The configuration of FIG. 1 is shown as including an intelligent processor **26**. Although the basic features of the invention may be implemented without the intelligent processor, the intelligent processor allows many enhanced features to be offered.

SSP **18** may utilize intelligent processor **26** to perform one or more specialized functions for interfacing with the caller at telephone station **14**. For example, if SCP **24** instructs SSP **18** to collect a voice print of the caller that will be used to determine the caller's identity, SSP **18** will, in some instances, use intelligent processor **26** to collect and analyze the print. In those instances, intelligent processor **26** collects a voice print, uses the print to generate an indication of the caller's identity, provides caller's identity to the SCP, and awaits further call processing instructions.

Intelligent processor **26** may also provide a video signal generation capability that is useful for processing video telephone and/or multimedia calls. For example, when a caller at telephone station **14** places a video call to telephone station **10**, SSP **18** detects that telephone station **10** is busy on another call and queries SCP **24** for call processing instructions. SSP **18** then forwards an indication that the call is a video call, an indication of a busy condition, the called telephone number, and the calling telephone number to SCP **24**. SSP **18** and SCP **24** process the call as described above, but in addition provide a video signal to telephone station **14**—the video signal being indicative of the forwarding destination of the call. In one possible embodiment, SCP **24** specifies to SSP **18** the type of video signal to be provided, and SSP **18** requests the appropriate video signal from intelligent processor **26**. Intelligent processor **26** then plays the video signal to telephone station **14**.

One skilled in the art will appreciate that various modifications can be made to the described embodiments without departing from the scope of the invention. For example, the calls between telephone stations **10** and **12** and between telephone stations **10** and **14** could be routed through other network elements, such as additional IXC switches. Also, the intelligent call forwarding service of the invention could be provided for calls to or from a mobile telephone, such as a cellular phone, or via cable television facilities. For example, a call placed from mobile telephone **28** (see FIG. **1**) via a commercially available mobile telephone switch **30** could be routed through switches **20** and **22** to SSP **18** for processing as described above.

We claim:

1. A call forwarding method, comprising the steps of:

a) coupling a call directed to a first telephone station to a service switching point, said call being initiated by a caller at a video telephone station;

b) using said service switching point to detect that said first telephone station is in a busy condition;

c) generating a query at said service switching point in response to said busy condition and sending said query from said service switching point to a service control point; and

d) responding to said sending of said query by:

(i) determining a forwarding destination for said call;

(ii) using an intelligent processor to play a video signal that is indicative of said forwarding destination to said video telephone station; and

(iii) forwarding said call to said forwarding destination.

2. A call forwarding method according to claim **1**, wherein the step of determining a forwarding destination comprises the steps of:

determining a number for said first telephone station; and

determining a forwarding destination based on said number.

3. A call forwarding method according to claim **1**, wherein the step of determining a forwarding destination comprises the steps of:

determining a time of day; and

determining a forwarding destination based on said time of day.

4. A call forwarding method according to claim **1**, wherein the step of determining a forwarding destination comprises the steps of:

determining a day of the week; and

determining a forwarding destination based on said day of the week.

5. A call forwarding method according to claim **1**, wherein the step of determining a forwarding destination comprises the steps of:

(i) playing an announcement to said caller and collecting a personal identification number from said caller; and

(ii) determining a forwarding destination for said call based on said personal identification number.

6. A call forwarding method according to claim **1**, wherein the step of determining a forwarding destination comprises the steps of:

(i) playing an announcement to said caller and collecting a voice print from said caller; and

(ii) using said intelligent processor to analyze said voice print and determine a forwarding destination based on said voice print.

7. A call forwarding method, comprising the steps of:

a) coupling a call directed to a first telephone station to a service switching point, said call being initiated by a caller at a video telephone station;

b) using said service switching point to detect that first telephone station is in a ting-no answer condition;

c) generating a query at said service switching point in response to said ring-no answer condition and sending said query from said service switching point to a service control point; and

d) responding to said sending of said query by:

(i) determining a forwarding destination for said call;

(ii) using an intelligent processor to play a video signal that is indicative of said forwarding destination to said video telephone station; and

(iii) forwarding said call to said forwarding destination.

8. A call forwarding method according to claim **7**, wherein the step of determining a forwarding destination comprises the steps of:

determining a number for said first telephone station; and

determining a forwarding destination based on said number.

9. A call forwarding method according to claim **7**, wherein the step of determining a forwarding destination comprises the steps of:

determining a time of day; and

determining a forwarding destination based on said time of day.

10. A call forwarding method according to claim 7, wherein the step of determining a forwarding destination comprises the steps of:

determining a day of the week; and

determining a forwarding destination based on said day of the week.

11. A call forwarding method according to claim 1, wherein the step of determining a forwarding destination comprises the steps of:

(i) playing an announcement to said caller and collecting a personal identification number from said caller; and

(ii) determining a forwarding destination for said call based on said personal identification number.

12. A call forwarding method according to claim 1, wherein the step of determining a forwarding destination comprises the steps of:

(i) playing an announcement to said caller and collecting a voice print from said caller; and

(ii) using said intelligent processor to analyze said voice print and determine a forwarding destination based on said voice print.

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